IN THE UNITED STATES DISTRICT COURT
FOR THE DISTRICT OF DELAWARE

MICROSOFT CORPORATION,	)	
	)	
Plaintiff,	)	
	)	
v.	)	C.A. No. 07-090 (SLR)
	)	
ALCATEL-LUCENT ENTERPRISE and GENESYS TELECOMMUNICATIONS LABORATORIES, INC.,	)	**REDACTED – PUBLIC VERSION**
	)	
Defendants.	)	

DEFENDANTS' REPLY BRIEF IN SUPPORT OF THEIR MOTION FOR SUMMARY JUDGMENT OF NON-INFRINGEMENT AND INVALIDITY OF ALL ASSERTED CLAIMS OF UNITED STATES PATENT NO. 6,430,289

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TABLE OF CONTENTS

TABLE OF AUTHORITIES

INTRODUCTION AND SUMMARY OF ARGUMENT

Defendants Alcatel Lucent Enterprise ("ALE") and Genesys Telecommunications Laboratories, Inc. ("Genesys") submit this reply brief in support of Defendants' Motion for Summary Judgment of Non-Infringement and Invalidity of All Asserted Claims of United States Patent No. 6,430,289 ("the '289 Patent") ("Defendants' '289 Opening Br.") (D.I.162).

Microsoft's entire infringement argument is that running a soft phone application constitutes "computer activity" within the meaning of the '289 Patent claims. However, the '289 Patent claims require more than mere computer activity – they require that the accused systems "*monitor*[] activity of the user computer" as a factor in determining when the user is available to take the call and then facilitate connecting the call to the user. (*See* Ex. 1 ('289 Patent) at 18:48-50, 18:55-65.) [redacted]

[redacted] Calls have been routed on that basis for decades. Microsoft cannot substitute routing based on telephone extension state for routing based on monitoring the user's computer activity because, among other things, it is entirely inconsistent with the premise of the '289 Patent. According to the description of the claimed invention, the point of monitoring a user's computer activity is to determine the user is present near the phone (and is thus more likely ***available*** to take a call). The inventions described and claimed in the '289 Patent use computer activity as a proxy for telephone availability. Simply put, if a user is typing on the keyboard or moving a mouse the '289 Patent presumes the user is available to take a call.

[redacted] That is not the invention described in the '289 patent (as the ALJ in the ITC case recognized) – it is the exact opposite.

The same is true with respect to the accused Genesys system. Microsoft points to no evidence showing the accused system routes agent interactions based on monitored activity of a user computer. Instead, Microsoft argues the call center agent's designated status (as either "busy" or "not busy") is good enough because if the agent indicates he or she is busy on an email chat and he or she is really not, the agent will get fired. Microsoft's argument simply highlights the fact that under its infringement scenario, computer activity is *assumed*, not monitored. The status of the agent is not the same thing as monitoring computer activity. Furthermore, the agent is not even the called party in the call center scenario because the caller does not call any particular agent. Therefore, the "agent" is not even the "user" under the claims of the '289 Patent.

Finally, Microsoft's arguments regarding the validity of the '289 Patent fail because it does not rebut expert testimony (and the Commission's opinion in the ITC case) that the telecommute server in Chestnut exists on both the telephone and computer networks. Additionally, Microsoft fails to refute explicit statements in Chestnut disclosing "monitoring activity of a user computer" connected to the computer network under Microsoft's construction of the claims.

ARGUMENT

A. Summary Judgment of the Accused ALE Systems' Non-Infringement of the '289 Patent Should be Granted

1. The Accused Systems Do Not Route Calls Based Upon the "Monitored Activity of the User's Computer"

Microsoft argues that being on a soft phone call entails user computer activity in the form of applications running on the computer (e.g., compressing, packetizing, and sending packets containing voice information over the LAN, etc.) (*See* Plaintiff Microsoft's Opposition to Defendants' Motion for Summary Judgment of Non-Infringement and Invalidity For All Asserted Claims of U.S. Patent No. 6,430,289 ("Plaintiff's '289 Opposition Br.") (D.I.185) at 13-18.) Microsoft, however, provides no evidence that the accused ALE systems ***monitor*** any of this so-called computer activity or route incoming calls ***based upon*** any monitored result of this alleged computer activity. [redacted]

a. The Accused Systems Do Not Infringe the Asserted Claims

Microsoft offers no evidence that the accused ALE systems process calls based upon any information beyond the rules set by the user and the state of the user's telephone extension. Indeed, there is nothing in the accused systems that provides for routing of calls based upon user computer activity. (*See* Ex. 2 (Hyde-Thomson ITC Hrg. Tr.) at 1347:2-1347:21.) The best Microsoft can do is point to the fact that VoIP soft phones are telephones that run on a computer. (*See* Declaration of William H. Beckmann in Support of Microsoft's Opposition to Defendant's '289 Motion for Summary Judgment ("Beckmann Decl.") (D.I.186) at ¶14.) ALE does not dispute this. However, what is missing is any evidence that the accused systems monitor computer activity when the user employs a soft phone (or any other time) in order to filter and route calls.

Microsoft's expert, Dr. Beckmann, spends a paragraph in his latest declaration explaining how a soft phone utilizes computer resources. (*Id.*) But Dr. Beckmann never attempts to explain how the accused systems monitor any of the computer activity he associates with running a soft phone application, much less how the accused systems route calls using any such information. (*Id.*) Instead, Dr. Beckmann makes the unsupported and inaccurate leap that because the OXE and OXO systems monitor "whether the called user is currently busy on another call" they must be monitoring computer activity if the user is using a soft phone for the user's telephone extension. (*Id.* at ¶¶18, 27.) Dr. Beckmann assumes, without evidence, that because the soft phone application uses computer resources to initiate a call, the accused systems must monitor that computer activity in processing incoming calls. (*See id.* at ¶18). However, Microsoft offers no evidence that the accused ALE systems process incoming calls based on monitoring any of the "soft phone" computer activity identified by Dr. Beckmann – the user's voice spoken into a computer microphone, mouse movements, keyboard input or digital data packets.

[redacted]

[redacted] (*See id.* at 1089:8-18, 1093:15-1095:5.)[1]

b. Microsoft's Infringement Scenario Fails

Microsoft's error is highlighted by the fact that monitoring when a user is busy on a soft phone call results only in a determination that the second party is ***not available*** – a direct contradiction to the purpose of the invention and the language of the claims. (*See* Ex. 16 (Final Initial and Recommended Determinations (January 28, 2008)) at 174, 180.) Monitoring when a user is on a soft phone demonstrates that the user is not available to take a call; however, the '289 Patent requires monitoring activity that indicates that the user is ***available*** to take a call.[2] Microsoft provides no example of monitored activity of the user computer being used to direct an incoming call through to the called party's phone. Instead, all its examples of alleged computer activity cause the call to be ***directed away from the user to another extension***. (*See, e.g.*, D.I.185 (Plaintiff's '289 Opposition Br.) at 5-6 ("forward on busy" feature of My Assistant), 6 ("OTUC will also determine whether the called party is currently busy on a call"), 17 ("a computer on which the user is running his softphone software application would be in a 'busy' state when engaged in a VoIP call"), and 17 ("a computer would be in an 'active' state when engaged in a VoIP call").

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1 [redacted]

2 A user not being on a soft phone no more indicates that the user is ***available*** to take the call than if a person is not on a traditional phone. This is exactly what the '289 Patent seeks to avoid – a situation where a person is not on a call but is also not available to take a call (because, for instance, the person is not physically near the phone). A call made in such instance will not result in a connection

Microsoft attempts to fill this gaping hole in its infringement allegation by arguing that a call can get through to the user if the user is ***not*** on the soft phone (D.I.185 (Plaintiff's '289 Opposition Br.) at 13.) In other words, Microsoft argues that the ability of the accused systems to direct calls to the dialed extension infringes. Aside from failing to explain how it can claim coverage of such long standing functionality when the background section of the patent acknowledges this as prior art (Ex. 1 ('289 Patent) at 1:17-30), Microsoft does not identify the monitored computer activity it alleges result in the routing of the call to the dialed extension in this example.

Furthermore, the example offered by Microsoft in an attempt to show that its alleged monitoring of user computer activity does not always result in a determination that the user is unavailable for a call does not satisfy the claim either. The absence of a determination that the second party is not available is not the same as determining ***when*** such party is available. As the '289 specification notes, a user may be off the phone, but out of the office, in which case the call will go unanswered because the user was not, in fact, available. (Ex. 1 ('289 Patent) at 1:23-50.) Indeed, it was exactly this problem that the inventor was trying to solve with the use of the monitored activity of the user computer, by using such information to determine that the user was physically present in the office to take a call. (Ex. 2 (Hyde-Thomson ITC Hrg. Tr.) at 1328:12-1330:2 (determining when the second party is available to take a call originated by the first party is very important to the problem the inventor was trying to solve).)

2. The Accused Systems Do Not Meet the '289 Patent's Requirement of Receiving an Indication of the Desire by a First Party to Set Up a Call with a Second Party

Microsoft's opposition to ALE's demonstration that the accused systems do not meet the '289 Patent's requirement of "receiving information from the telephone network that a first party from whom a call is originating desires to establish telephone communication with a second

party" is nothing more than a rehash of its argument against ALE's claim construction. (*See* Ex. 5 (Beckmann Dep. Tr.) at 227:8-230:18 (admitting his opinions are based on Microsoft's claim construction and that he offers no opinion under the ALE construction).)

Microsoft claims that ALE's argument concerning this element is simply an attempt to read limitations from the specification into the claims. Microsoft relies on Federal Circuit case law noting that claims need not disclaim ***every*** feature of prior art discussed in the Background section. (D.I.185 (Plaintiff's '289 Opposition Br.) at 23, citing *Ventana Med. Sys., Inc. v. BioGenex Labs, Inc.*, 473 F.3d 1173, 1180 (Fed. Cir. 2006).) This undisputed law is unavailing to Microsoft because, as explained more fully in ALE's Markman response brief, ALE is not reading limitations into the claims. (Defendants' Answering Claim Construction Brief (D.I.192) at 28.) Rather, ALE has construed the explicit claim language in light of the specification's description of the problem the '289 Patent inventor purported to solve – namely, that having to make a call to determine if the other party is available is a waste of resources.

The specification explains that the claimed invention solves the problem of providing the caller "no choice but to place a call to the destination telephone and hope that the callee answers" because the prior art systems were incapable of determining *when* a particular callee was actually available to take a call. (Ex. 1 ('289 Patent) at 1:31-35; *see also id.* 1:38-43 (failed attempts to contact a party, and/or numerous calls back and forth with the parties leaving messages in an effort to set up a call was a waste of resources).) Placing a telephone call to determine ***whether*** the other party is available to take a call, as Microsoft's construction requires, would undercut the entire stated purpose of the invention. *See Phillips v. AWH Corp.*, 415 F.3d 1303, 1316 (Fed. Cir. 2005) (quoting *Renishaw PLC v. Marposs Societá Per Azion*, 158 F.3d 1243, 1250 (Fed. Cir. 1998) ("Ultimately, the interpretation to be given a term can only be determined and confirmed

with a full understanding of what the inventors actually invented and intended to envelop with the claim.")).

[redacted] (*See* D.I.185 (Plaintiff's '289 Opposition Br.) at 22-25; *see also* Ex. 5 (Beckmann Dep. Tr.) at 227:8-230:18.)

3. [redacted]

Microsoft argues at length that Defendants misread this claim when Defendants suggest that being busy on a soft phone call can never indicate when a called party is available. The reason, Microsoft says, is that "availability" is a separate concept from the "monitored activity of a user computer." In fact, Microsoft argues that the accused systems monitor the activity of the user computer by "consider[ing] whether the user is already 'busy' using [a soft phone]." (D.I.185 (Plaintiff's '289 Opposition Br.) at 27.) Microsoft is wrong.

The claim requires using "monitored activity of the user computer of the second party to determine when the second party is available to take the call." (Ex. 1 ('289 Patent) at 18:55-61.) This expressly requires the monitoring of the user computer's activity to provide a determination that the party is available. Microsoft's claimed infringing configuration simply does not do this.

Even if "consider[ing] whether the user is already 'busy' using [a soft phone]" can be considered "monitoring activity of the user computer," the fact remains that such monitoring cannot determine that the party is ***available***, as the limitation requires. For instance, if the second party is not busy on the soft phone, Microsoft's example would deem the second party "available." But, of course, we know absolutely nothing about the second party's availability except that we cannot confirm that the party is not available. The party could be away from the computer altogether – the soft phone would not be busy, but the party clearly would not be available. All that can be gathered from "monitoring" whether the user is already busy on a call is that, if the user is busy on a soft phone call, then that user is not available. [redacted]

Microsoft desperately attempts to change an invention directed to assessing whether someone is available to take a call by using computer activity as a proxy for availability into one where monitoring any activity of any type for any purpose infringes regardless of whether a call is routed based on that activity or whether such activity indicates availability to take a call. Cherry picking claim construction maxims does not change the claimed invention, the described invention or the fact that the accused systems do not practice the invention claimed and described in the '289 Patent.

4. The Accused Systems Do Not Indirectly Infringe the '289 Patent

For the reasons set forth in the concurrently filed Reply Brief in Support of Its Motion for Summary Judgment of Non-Infringement and Invalidity of All Asserted Claims of United States Patent No. 6,421,439 at Section II.A.4 (filed concurrently and incorporated herein by reference),

Microsoft has failed to meet its burden to show contributory infringement of the accused ALE systems because the accused systems have substantial non-infringing uses.

B. Summary Judgment of Non-Infringement of the Accused Genesys System is Warranted

1. [redacted]

(Ex. 1 ('289 Patent) at 1:33-35.)

[redacted] (*Id.*; *see also* Ex. 7 (Expert Report of Dr. Leonard J. Forys, Ph.D., Regarding Genesys Telecommunications Laboratories, Inc.'s Non-Infringement of U.S. Patent No. 6,430,289 in Rebuttal to the Expert Report of Dr. William H. Beckmann ("Forys Rebuttal Report")) at ¶84.) Microsoft fails to point to any evidence to the contrary showing that the accused Genesys system routes interactions based on any monitored activity and therefore no factual dispute exists.

2. Microsoft Fails to Show that the Accused Genesys System Monitors "the Activity of a User Computer" and Thus No Factual Dispute Exists to Preclude Summary Judgment.

a. The Fact that a User is Assigned a Chat Does Not Mean the User is Active on the Computer and Thus Unavailable

[redacted] (D.I.185 (Plaintiff's '289 Opposition Br.) at 2, 35.) However, the asserted claims require that the "monitored activity of the user computer [be used] to determine when the second party is available to take the call originated by the first party." Thus, Microsoft is attempting to rewrite the claim to use the activity of the user[3] itself (and not the activity of the user's computer) as a proxy that the user is active on the computer and thus unavailable to take a call.

[redacted] (Ex. 1 ('289 Patent) at 14:30-49; D.I.186 (Beckmann Decl.) at ¶34.)

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3 Once again, the agent is not the called party in the call center environment and thus does not meet the "user" limitation of the claims, but for purposes of this motion only, ALE will accept Microsoft's misapplication of this limitation.

[redacted] (Ex. 9 (Forys Dep. Tr.) at 271:4-273:20.)

b. The Accused Genesys System Does Not Perform Automatic Monitoring

In distinguishing U.S. Patent No. 5,999,965 ("Kelly"), Dr. Beckmann explicitly cited to the section in Kelly discussing automatic monitoring of the agent's statuses (Ex. 10 ('965 Patent) at 19:24-27 ("Next, the agent thread monitors the connection for status packets indicating the status of the agent.").) Thus, Dr. Beckmann conceded that "monitoring activity of a user's computer" means that the accused Genesys system must be proactively checking or polling the user's actions to be considered "monitoring." This is reinforced by Dr. Beckmann's statement that "the agent is responsible in the Kelly system for transmitting the packets, they reflect the agent's activity-not the computer's." (Ex. 11 (Second Expert Report of William Beckmann ("Beckmann Rebuttal Report")) at 83.) Without the computer performing some frequent check

[4] Microsoft states that "Dr. Forys revealed that that opinions were in part based on an inspection of a system specially configured by him" and thus argues summary judgment is inappropriate because it lacks essential facts. (D.I.186 (Plaintiff's '289 Opposition Br.) at 33, n.4.) Genesys provided Microsoft and Dr. Beckmann with multiple demonstrations of the accused Genesys system allowing them full right to configure the system as they sought. For Microsoft to argue now that it lacks the facts necessary for it to defend a summary judgment motion is not accurate. [redacted]

or polling while the agent is typing, there would never be any computer activity – only agent activity – even though there is a software program actively running and receiving the text.

[redacted] (D.I.186 (Beckmann Decl.) at ¶34.) Microsoft refers to the "Handling" statistic to also support that the accused Genesys system performs automatic monitoring. (D.I.185 (Plaintiff's '289 Opposition Br.) at 34.) [redacted] Thus, contrary to Microsoft's assertion, the statistic does not reflect "the fact that the agent and his computer are occupied," but only reflects whether a chat has started or ended.

3. The Accused Genesys System Does Not "Receiv[e] Information From the Telephone Network That a First Party From Whom a Call is Originating [(i.e., Caller)] Desires to Establish Telephone Communication With a Second Party"

The asserted claims require "at the computer network, receiving information from the telephone network that a first party from whom a call is originating desires to establish telephone communication with a second party." (Ex. 1 ('289 Patent) at 18:44-47, 19:32-35.) Microsoft incorrectly asserts that the accused Genesys system meets this limitation under Microsoft's construction because the T-Server does not know who the second party will be at the time it receives information about the call. (D.I.185 (Plaintiff's '289 Opposition Br.) at 36).[5] [redacted]

5 This argument by Microsoft illustrates the problem with its contention that the call center agent is the "user" in the recited claim limitation. Because the caller does not dial any particular call center agent, the agent cannot be the "user" of the claims.

[redacted] (*See id.* at ¶69 ("call center agents do not have their own telephone numbers").) This is in direct contrast with the claim limitation's requirement that the second party be known when the call is received at the computer network.

Additionally, Microsoft provides no opinion on infringement under Genesys' proposed construction of this term and ignores the fact that the '289 Patent is directed toward a system to improve upon the deficient prior art systems. (Ex. 1 ('289 Patent) at 1:33-35.) [redacted] (D.I.156 (Forys Decl.) at ¶25.) Accordingly, under either construction there is no material dispute that this limitation is not met.

4. No Indirect Infringement

Microsoft cannot establish that there is indirect infringement because specific optional modules, such as chat or e-mail, along with Inbound Voice are required to satisfy the '289 claim limitations. (D.I.162 (Defendants' '289 Opening Br.) at 16; *see also* Ex. 7 (Forys Rebuttal Report) at ¶31) [redacted] Microsoft has not established that any customer has

ever purchased the specific optional modules along with Inbound Voice to form the alleged infringing accused Genesys system.

In addition, with respect to method claims 1 and 3, Microsoft cites numerous documents, but those documents all lack actual evidence showing any Genesys customers practicing the accused Genesys system. (D.I.185 (Plaintiff's '289 Opposition Br.) at 37.) [redacted]

C. The '289 Patent is Invalid in View of the Chestnut Patent Under Microsoft's Construction

1. Chestnut Discloses "Receiving Information From the Telephone Network that a First Party From Whom a Call is Originating Desires to Establish Telephone Communication With a Second Party"

Microsoft attempts to manufacture a genuine issue of fact by asserting ALE never argued this element was disclosed by the Chestnut Patent (U.S. Patent No. 6,041,114). This is completely inaccurate. In Mr. Hyde-Thomson's declaration in support of Defendants' opening brief for summary judgment he clearly outlines how this limitation is met: "Chestnut teaches that the telecommute server receives the call when the telecommute server 'intercepts' the call." (Ex. 13 (Expert Report of Mr. Henry Hyde-Thomson Regarding Invalidity and Materiality (Corrected) ("Hyde-Thomson's Opening Report")) at ¶¶381-384; *see also id.* at Ex. H, 1(e); D.I.161 (Hyde-Thomson Decl.) at ¶98.) Dr. Beckmann attempts to refute Mr. Thomson's argument, but he fails to understand that the telecommute server can be on both the telephone and computer network and thus receive information at the computer network and satisfy this

limitation. (D.I.186 (Beckmann Decl.) at ¶41; *see also* Ex. 14 (Commission Decision (June 6, 2008)) at 26 (finding "that Chestnut discloses that the telecommute server exists on both the telephone and computer networks").)

2. Chestnut Discloses Monitoring Activity of a User Computer Connected to the Computer Network

The Chestnut Patent monitors whether or not a user is ***logged on*** to a computer network. According to Microsoft, being logged on is not a "status" (Microsoft's construction) or "activity" (ALE's construction) of a user computer, but is a "precursor" to activity of a user computer. (D.I.185 (Plaintiff's '289 Opposition Br.) at 38-39 (citing D.I.186 (Beckmann Decl.) at ¶42.))

In contrast to Microsoft's unsupported position, ALE cites an explicit statement from the Chestnut Patent showing that Chestnut discloses forwarding a telephone call based on the user's status as logged onto the computer network. (Ex. 15 (Chestnut Patent) at 2:34-44; D.I.162 (Defendants' '289 Opening Br.) at 24-25.)

[redacted]

3. Chestnut Discloses "Using the Set of Pre-Determined Rules . . . to Determine When the Second Party is Available to Take the Call"

Microsoft invents an "inconsistency" to obscure the fact that Chestnut discloses this limitation under Microsoft's application of the claims. Microsoft argues that Defendants believe the "telecommute server" is on the computer network in their '289 briefing, but located on the telephone network in their '439 briefing. This is wrong. Defendants clearly state that "[i]n Chestnut, the 'telecommute server' is connected to both the telephone network and the computer

network and is part of both networks." (Defendant ALE's Opening Brief in Support of Its Motion for Summary Judgment of Non-Infringement and Invalidity of All Asserted Claims of United States Patent No. 6,421,439 ("the '439 Patent") ("ALE's '439 Opening Br.") (D.I.160) at 28.) Once it is understood that the telecommute server is on both networks, Microsoft's illusory argument falls away.

The Chestnut Patent explicitly discloses pre-determined rules that are stored on the computer network portion of the telecommute server to determine when the called party is available to take a call. For example, in the Chestnut Patent, a call will be forwarded to a party based on whether the party has indicated that he is available at a certain time (Ex. 15 (Chestnut Patent) at 5:18-26 ("[t]he telecommute server 2, ***can also forward incoming calls based upon other criteria including day or date, time of day***, the identity of the caller, or any preprogrammed set of rules.") (emphasis added); *see also* Ex. 14 (Commission Decision (June 6, 2008)) at 25 (adopting for the '439 Patent the "ALJ's findings that Chestnut discloses . . . 'data structure contained within a computer network to store user-selectable criteria for call processing'").)

4. Chestnut Discloses "Storing a Set of Predetermined Rules for Determining When the Second Party is Available to Take a Call from the First Party"

Microsoft relies on the same nonexistent "inconsistency" in connection with the telecommute server with respect to this claim limitation. Thus, Microsoft's argument fails for the same reasons as it did for the previous element. Moreover, Microsoft ignores that the Chestnut Patent discloses the storing of pre-determined rules on the computer network for determining when a party is available to take a call. (*See* Ex. 15 (Chestnut Patent) at 5:13-26; Fig. 1, 4:36-57.)

5. Chestnut Discloses "Using the Information Processed at the Computer Network to Facilitate Connecting the Call Originated by the First Party Through the Telephone Network to the Second Party"

Microsoft does not oppose Defendants' argument that the Chestnut Patent discloses the limitation of claims 1 and 8 (not present in claim 7) that provide "using the information processed at the computer network to facilitate connecting the call originated by the first party through the telephone network to the second party." (Ex. 1 ('289 Patent) at 18:62-65.)

6. Chestnut Discloses a "Computer Program Product"

Microsoft asserts that Defendants have failed to point to a computer program product performing a "series of steps." (D.I.185 (Plaintiff's '289 Opposition Br.) at 40.) Microsoft has mistaken the Defendants' burden. Defendants are required to identify a computer program product in Chestnut and show the other claim limitations are disclosed. The Defendants have met their burden.

Chestnut describes "CTI applications" (i.e. a software program) and Mr. Hyde-Thomson explained that "the Chestnut patent includes the idea of a computer program product. [redacted]

(Ex. 2 (Hyde-Thomson ITC Hrg. Tr.) at 1399:6-12.)

This CTI application and/or software program identified by Mr. Thomson are used to perform the other claim limitations found in claims 7, 8, and 10. However, those other claim limitations have already been shown to be disclosed in Chestnut. (*See* D.I.162 (Defendants' '289 Opening Br.) at 24-28.) Thus, Microsoft's argument that Defendants must identify a "series of steps" performed by the computer program product is without merit as it has already been shown.

CONCLUSION

For the foregoing reasons, ALE respectfully requests that the Court grant ALE's motion and enter summary judgment of non-infringement and invalidity of U.S. Patent No. 6,430,289.

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June 30, 2008

2389609

CERTIFICATE OF SERVICE

I, Maryellen Noreika, hereby certify that on July 7, 2008 I electronically filed the foregoing document, which will send notification of such filing(s) to the following:

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I also certify that copies were caused to be served on July 7, 2008 upon the following in the manner indicated:

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EXHIBIT 1

US006430289B1

(12) United States Patent
Liffick

(10) Patent No.: US 6,430,289 B1
(45) Date of Patent: Aug. 6, 2002

(54) SYSTEM AND METHOD FOR COMPUTERIZED STATUS MONITOR AND USE IN A TELEPHONE NETWORK

(75) Inventor: Stephen Mitchell Liffick, Seattle, WA (US)

(73) Assignee: Microsoft Corporation, Redmond, WA (US)

(*) Notice: Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

(21) Appl. No.: 09/291,693

(22) Filed: Apr. 13, 1999

(51) Int. Cl.[7] H04M 1/00
(52) U.S. Cl. 379/900; 379/142.15; 370/352
(58) Field of Search 379/201.06, 209.07, 379/201.08, 201.1, 210.11, 142.15, 196, 197, 198, 199, 900; 370/352, 353, 354

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* cited by examiner

Primary Examiner—Craighton Smith
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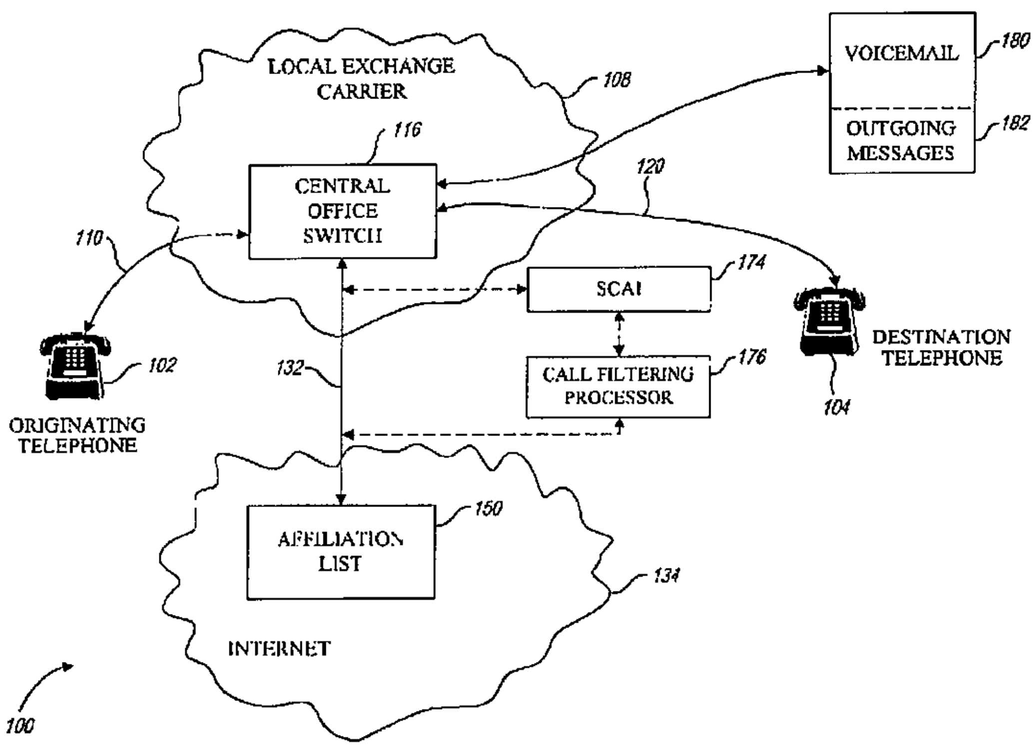
(57) ABSTRACT

A telecommunication system combines telephone technology and computer network technology to monitor a caller and callee's computer activity and to access call processing criteria selected by the caller and callee and stored on the computer network. A component of the telephone system, such as a central office switch, accesses the caller and callee call processing criteria. The system evaluates the call processing criteria and, when conditions for both caller and callee are met, the telephone system initiates a telephone call between the caller and callee. The call processing criteria may include accepting all calls, no calls, or calls only from specified parties. In addition, the call processing criteria can vary in accordance with the time of day or an individual's personal preferences, or status, such as when an individual is in a meeting. A user's computer activity may also be monitored and the computer status as idle or active may be reported to the computer network as part of the call processing criteria.

20 Claims, 10 Drawing Sheets



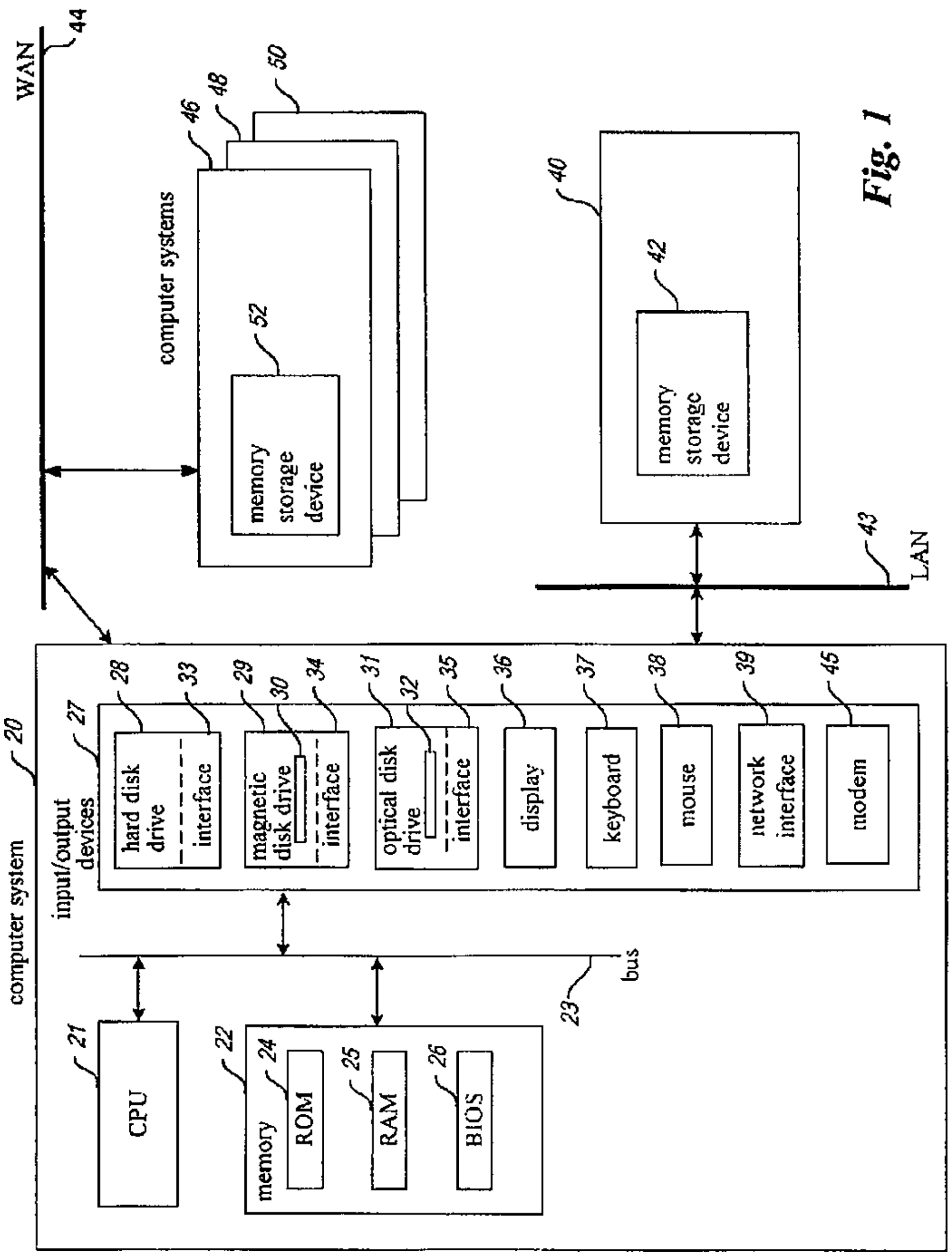
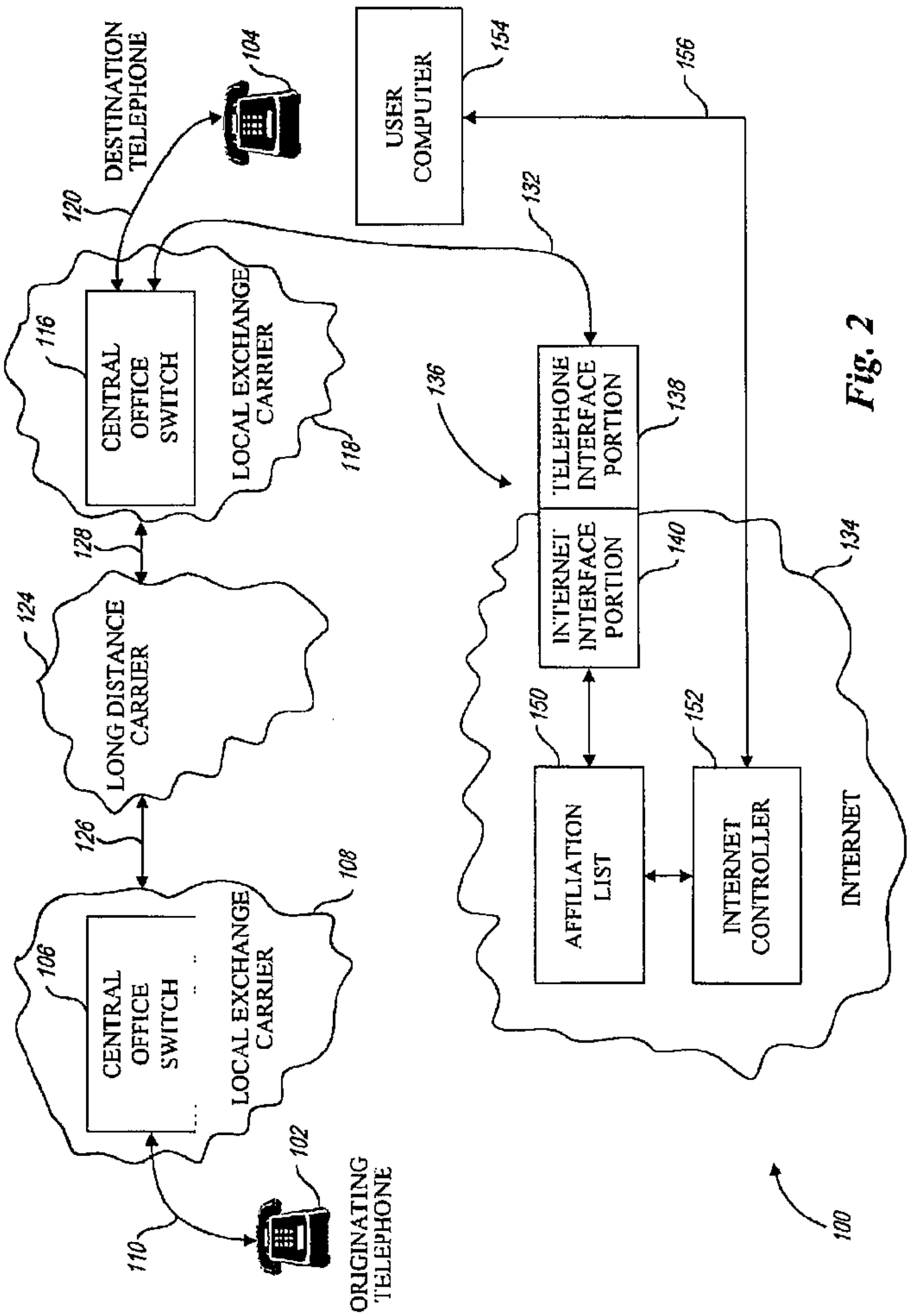

WAN
44
computer systems
46
48
50
52
memory storage device
40
42
memory storage device
43
LAN
computer system
20
27
input/output devices
28
hard disk drive
33
interface
29
30
magnetic disk drive
34
interface
31
32
optical disk drive
35
interface
36
display
37
keyboard
38
mouse
39
network interface
45
modem
bus
23
21
CPU
22
memory
24
ROM
25
RAM
26
BIOS

Fig. 1



DESTINATION TELEPHONE
104
USER COMPUTER
154
156
120
132
CENTRAL OFFICE SWITCH
116
LOCAL EXCHANGE CARRIER
118
136
TELEPHONE INTERFACE PORTION
138
INTERNET INTERFACE PORTION
140
134
128
LONG DISTANCE CARRIER
124
150
152
AFFILIATION LIST
INTERNET CONTROLLER
INTERNET
126
108
CENTRAL OFFICE SWITCH
106
LOCAL EXCHANGE CARRIER
102
110
ORIGINATING TELEPHONE
100

Fig. 2

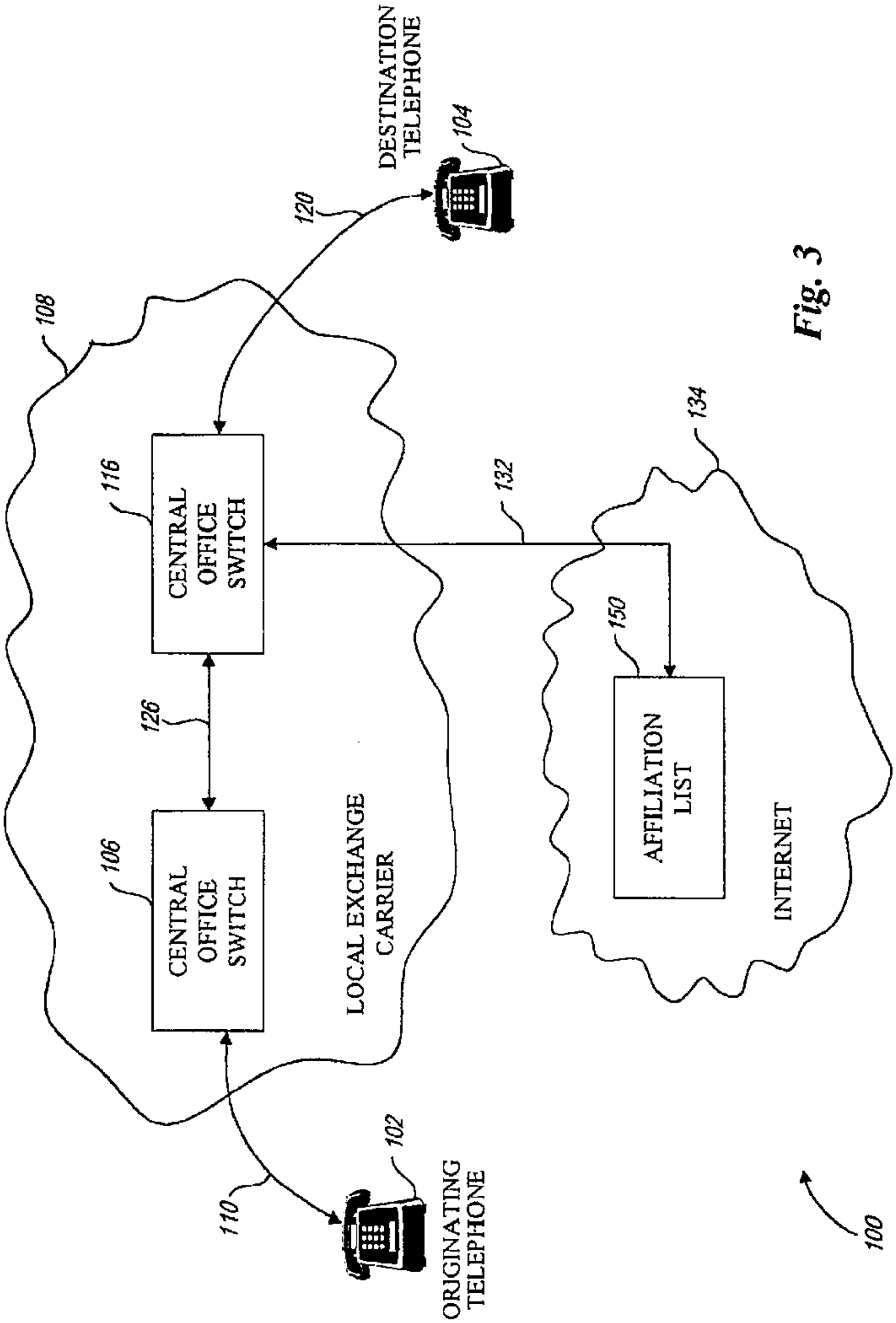
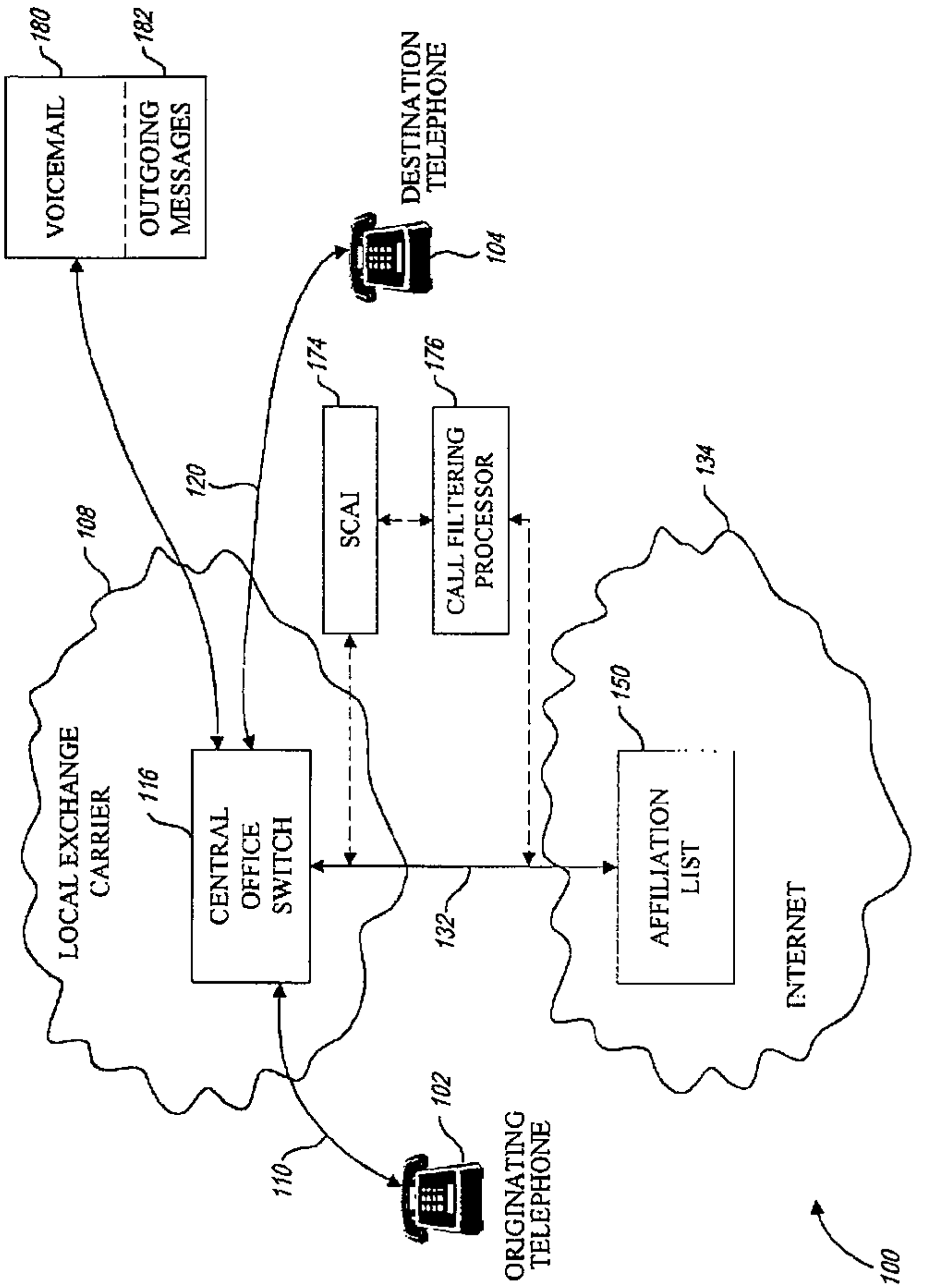

DESTINATION TELEPHONE
104
120
108
116
CENTRAL OFFICE SWITCH
132
134
150
AFFILIATION LIST
126
106
CENTRAL OFFICE SWITCH
LOCAL EXCHANGE CARRIER
INTERNET
102
110
ORIGINATING TELEPHONE
100

Fig. 3



VOICEMAIL
OUTGOING MESSAGES
180
182
DESTINATION TELEPHONE
104
174
176
SCAI
CALL FILTERING PROCESSOR
120
108
134
LOCAL EXCHANGE CARRIER
116
CENTRAL OFFICE SWITCH
150
AFFILIATION LIST
132
INTERNET
102
110
ORIGINATING TELEPHONE
100

Fig. 4

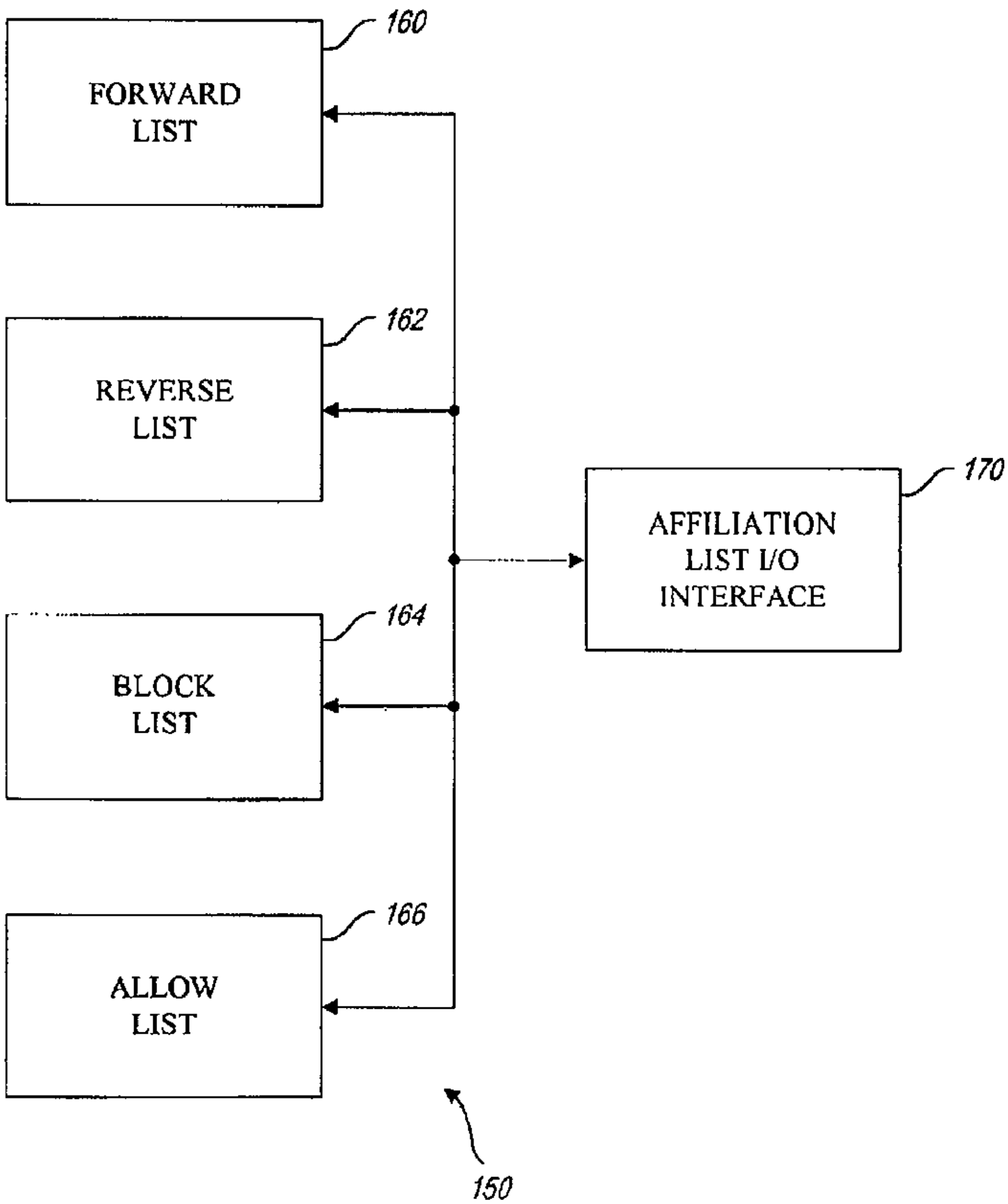

160
FORWARD LIST
162
REVERSE LIST
164
BLOCK LIST
166
ALLOW LIST
170
AFFILIATION LIST I/O INTERFACE
150

Fig. 5

Name	Bob Smith
Subscriber Name	bobxyz@msn.com
Phone 1	(425) 555-1234
Phone 2	(425) 555-1235
⋮	
Name	Jim Smith
Subscriber Name	NONE
Phone 1	(206) 555-1236
⋮	
Name	John Adams
Subscriber Name	johnxyz@aol.com
Email Alias	atom smasher xyz
Phone 1	(703) 555-1237
Phone 2	(703) 555-1238
Phone 3	(703) 555-1239

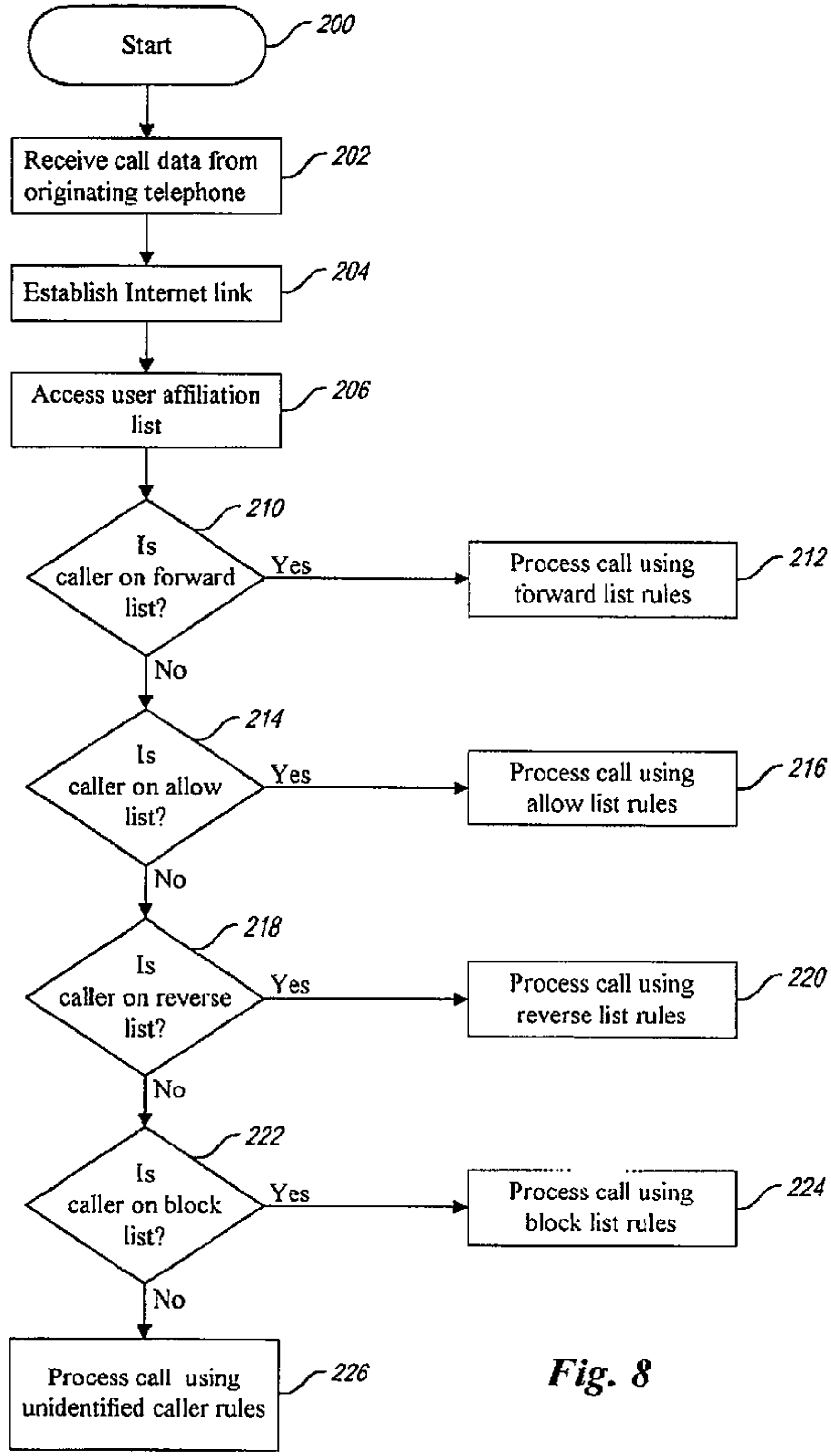
166

Fig. 6

Name		Bob Smith
Subscriber Name		bobxyz@msn.com
Phone 1		(425) 555-1234
Phone 2		(425) 555-1235
Status		Allowed
⋮		
Name		Jim Smith
Subscriber Name		NONE
Phone 1		(206) 555-1236
Status		Blocked
⋮		
Name		John Adams
Subscriber Name		johnxyz@aol.com
Email Alias		atom smasher xyz
Phone 1		(703) 555-1237
Phone 2		(703) 555-1238
Phone 3		(703) 555-1239
Status		Conditional
Phone 1	-	Allowed
Phone 2	-	Allowed 9:00 a.m. - 11:30 a.m.
Phone 3	-	Blocked

150

Fig. 7



Start
200
Receive call data from originating telephone
202
Establish Internet link
204
Access user affiliation list
206
Is caller on forward list?
210
Yes
No
Process call using forward list rules
212
Is caller on allow list?
214
Yes
No
Process call using allow list rules
216
Is caller on reverse list?
218
Yes
No
Process call using reverse list rules
220
Is caller on block list?
222
Yes
No
Process call using block list rules
224
Process call using unidentified caller rules
226

Fig. 8

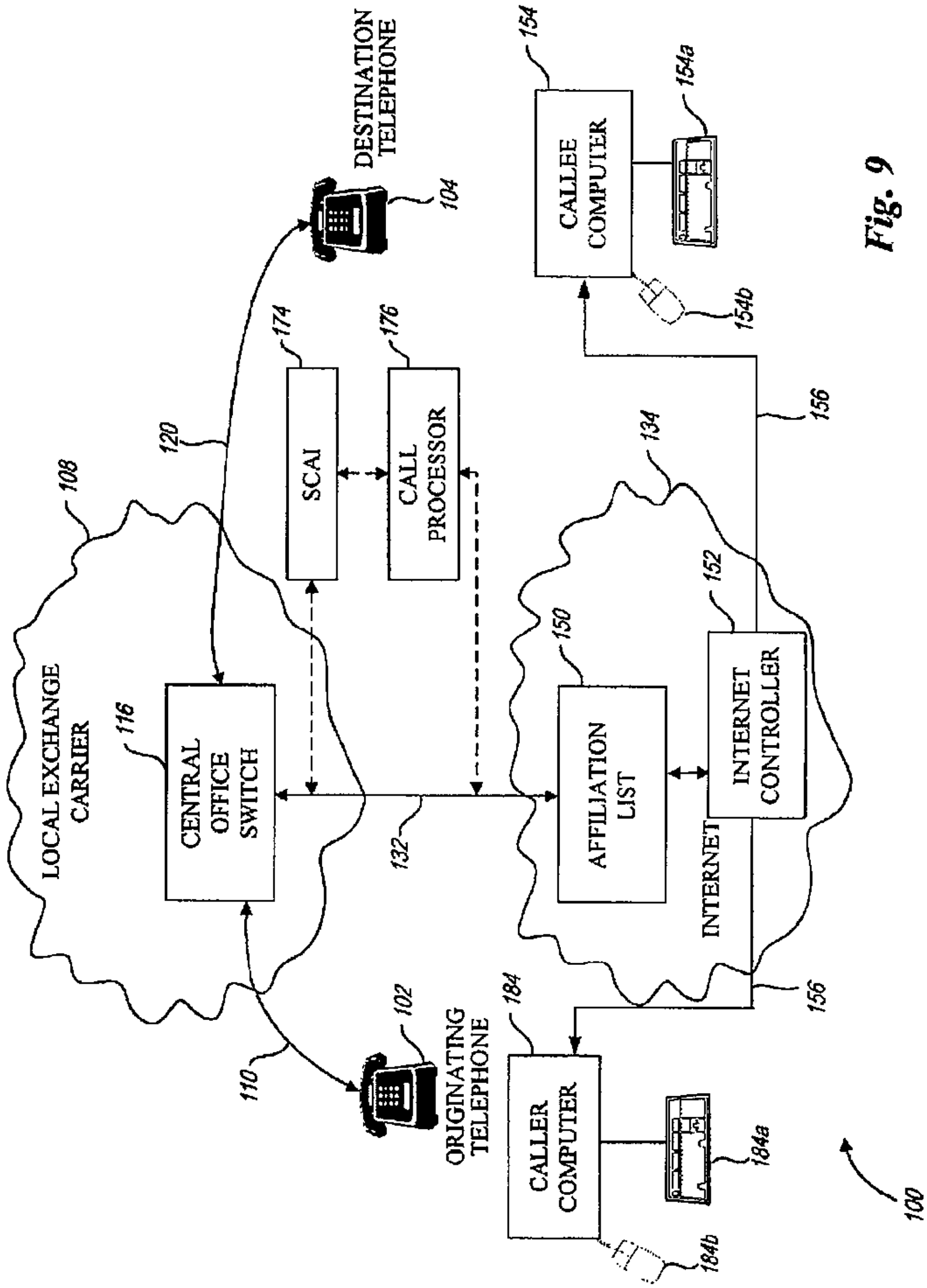
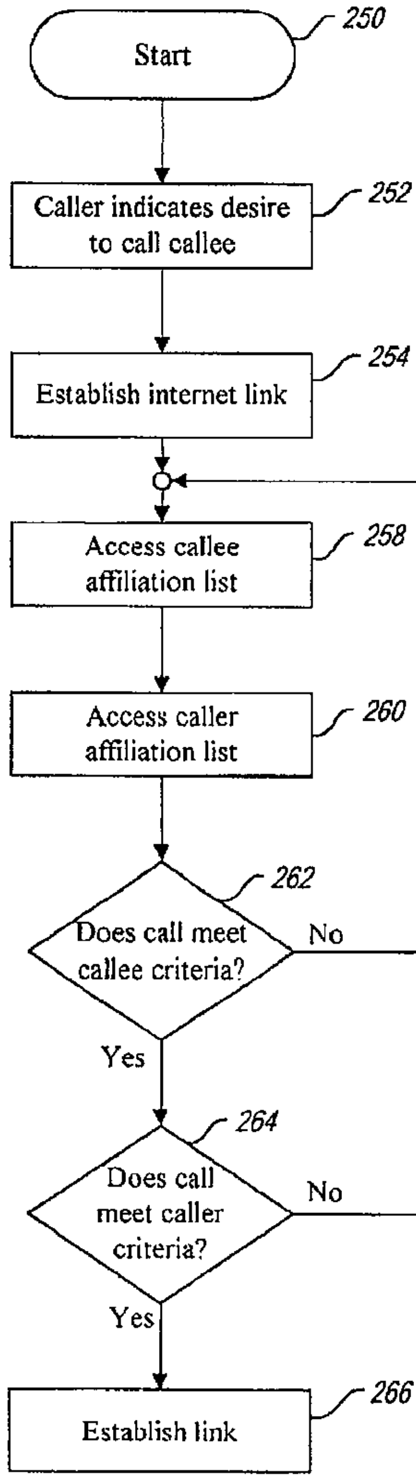

LOCAL EXCHANGE CARRIER
108
CENTRAL OFFICE SWITCH
116
120
110
ORIGINATING TELEPHONE
102
DESTINATION TELEPHONE
104
SCAI
174
CALL PROCESSOR
176
132
AFFILIATION LIST
150
INTERNET CONTROLLER
152
INTERNET
134
156
156
CALLEE COMPUTER
154
154a
154b
CALLER COMPUTER
184
184a
184b
100

Fig. 9



Start
250
Caller indicates desire to call callee
252
Establish internet link
254
Access callee affiliation list
258
Access caller affiliation list
260
Does call meet callee criteria?
262
No
Yes
Does call meet caller criteria?
264
No
Yes
Establish link
266

Fig. 10

SYSTEM AND METHOD FOR COMPUTERIZED STATUS MONITOR AND USE IN A TELEPHONE NETWORK

TECHNICAL FIELD

The present invention is directed generally to telecommunications and, more particularly, to a system and method for establishing a telephone communication link using status reporting information from an independent computer network.

BACKGROUND OF THE INVENTION

Telephone communication systems have increased in both size and complexity. Early telephone systems required a human operator to manually connect an originating telephone with a destination telephone. With the introduction of automatic switching technology, the need for human operators to connect each and every call disappeared. However, even automated switches did not provide the wide range of features available on most telephone systems, such as voicemail, caller identification, call waiting, call forwarding, three-way calling and the like. Most telephone systems today include these features and allow the customer to select one or more features to customize their telephone service. With features such as voicemail, the telephone switching system must recognize when the destination telephone is either busy or remains unanswered. If either of these conditions occur, the calling party is routed to the voicemail service associated with the destination telephone.

Despite these improvements, telephone systems are incapable of determining when a particular recipient (i.e., a callee) may be available to receive a call. The caller has no choice but to place a call to the destination telephone and hope that the callee answers. Alternatively, the caller may leave a voicemail indicating a specific time at which the caller will place yet another call. This is an undesirable activity since it requires multiple calls, thus utilizing telecommunication capabilities in an inefficient manner. In addition, repeated or failed attempts to actually reach the callee are a waste of human resources since the parties must often call back and forth to each other a number of times before actually reaching the desired party. Therefore, it can be appreciated that there is a significant need for a system and method that can establish a telephone communication link when both parties are available to communicate. The present invention provides this and other advantages as will be apparent from the following detailed description and accompanying figures.

SUMMARY OF THE INVENTION

A system to specify user-selectable criteria for call processing is implemented on a telephone system, such as a public switched telephone network (PSTN). The user-specified call processing criteria is stored on a network that is accessible by the user for data entry and/or editing, and is also accessible by the PSTN to determine whether call processing criteria exists for the particular caller. The Internet provides a readily available data structure for storage of the user-selectable call processing criteria. The user can establish a database stored on the Internet in association with the user's telephone number and indicating the user-selectable call processing criteria for one or more potential callers.

The caller may be identified by caller identification data, such as automatic number identification (ANI). Based on the destination telephone number and the caller identification data, the PSTN accesses the Internet and examines an affiliation list corresponding to the destination telephone number. If the caller identification data is present in the affiliation list, the call may be processed in accordance with the user-specified criteria for that particular caller.

Both the caller and callee can specify user-selectable call processing criteria. The potential callee can specify call processing criteria for all incoming calls, such as providing a list of individuals from whom the person will accept calls, a list of individuals from whom the person will not accept calls, or conditional criteria, such as accepting or blocking calls during certain times of day or during certain periods of activity, such as when the user may be otherwise occupied and unwilling to accept an incoming call. In addition, the potential callee's computer activity may be monitored and the status of the computer as idle or active may be reported to the computer network. The caller indicates a desire to establish a communication link with the callee. The computer network accesses the caller's call processing criteria and the callee's call processing criteria. The call processing criteria for both the caller and callee are analyzed and when all conditions are met, a telephone communication link is established between an originating telephone associated with the caller and a destination telephone associated with the callee.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 illustrates a computer system that includes components to implement the system of the present invention.

FIG. 2 is a functional block diagram outlining the operation of the present invention.

FIG. 3 is a functional block diagram of an alternate telecommunications configuration implementing the present invention.

FIG. 4 is a functional block diagram of another alternative telecommunications configuration implementing the present invention.

FIG. 5 is a functional block diagram providing details of the affiliation list of the system of FIG. 2.

FIG. 6 illustrates sample data provided in the list of FIG. 5.

FIG. 7 illustrates additional sample data provided in the list of FIG. 3.

FIG. 8 is a flowchart illustrating the operation of the system of FIG. 2.

FIG. 9 is a functional block diagram illustrating the system of the present invention to process a call in accordance with both a caller and callee call processing criteria.

FIG. 10 is a flowchart illustrating the operation of the system of FIG. 9.

DETAILED DESCRIPTION OF THE INVENTION

Existing telephone technology does not provide the telephone subscriber with a technique for controlling access to the user's telephone. Features such as caller ID identify the caller, but do not control access to the user's telephone. Thus, the conventional telephone system forwards the user to extreme options. The user may answer all incoming calls or may choose not to answer any incoming calls. However, the present invention provides selective options in between these two extremes. The present invention combines telephone technology with Internet technology to allow the user

to "filter" incoming calls based on user-selected criteria. In particular, the user may establish a series of lists, stored on the Internet in association with the user's telephone, to filter incoming calls and thereby control access to the user's telephone. In addition, it is possible to monitor the activity or status of both a caller and a callee and establish a communication link between the caller's telephone and the callee's telephone when status data indicates that both are available for a telephone call.

FIG. 1 and the following discussion are intended to provide a brief, general description of a suitable computing environment in which the invention may be implemented. Although not required, the invention will be described in the general context of computer-executable instructions, such as program modules, being executed by a personal computer. Generally, program modules include routines, programs, objects, components, data structures, etc. that perform particular tasks or implement particular abstract data types. Moreover, those skilled in the art will appreciate that the invention may be practiced with other computer system configurations, including hand-held devices, multiprocessor systems, microprocessor-based or programmable consumer electronics, network PCs, minicomputers, mainframe computers, and the like. The invention may also be practiced in distributed computing environments where tasks are performed by remote processing devices that are linked through a communications network. In a distributed computing environment, program modules may be located in both local and remote memory storage devices.

With reference to FIG. 1, an exemplary system for implementing the invention includes a general purpose computing device in the form of a conventional personal computer 20, including a processing unit 21, a system memory 22, and a system bus 23 that couples various system components including the system memory to the processing unit 21. The system bus 23 may be any of several types of bus structures including a memory bus or memory controller, a peripheral bus, and a local bus using any of a variety of bus architectures. The system memory 22 includes read only memory (ROM) 24 and random access memory (RAM) 25. A basic input/output system 26 (BIOS), containing the basic routines that helps to transfer information between elements within the personal computer 20, such as during start-up, may be stored in ROM 24.

The personal computer 20 further includes input/output devices 27, such as a hard disk drive 28 for reading from and writing to a hard disk, not shown, a magnetic disk drive 29 for reading from or writing to a removable magnetic disk 30, and an optical disk drive 31 for reading from or writing to a removable optical disk 32 such as a CD ROM or other optical media. The hard disk drive 28, magnetic disk drive 29, and optical disk drive 31 are connected to the system bus 23 by a hard disk drive interface 33, a magnetic disk drive interface 34, and an optical drive interface 35, respectively. The drives and their associated computer-readable media provide nonvolatile storage of computer readable instructions, data structures, program modules and other data for the personal computer 20. Although the exemplary environment described herein employs a hard disk, a removable magnetic disk 30 and a removable optical disk 32, it should be appreciated by those skilled in the art that other types of computer readable media which can store data that is accessible by a computer, such as magnetic cassettes, flash memory cards, digital video disks, Bernoulli cartridges, random access memories (RAMs), read only memories (ROM), and the like, may also be used in the exemplary operating environment. Other I/O devices 27, such as a display 36, keyboard 37, mouse 38, and the like may be included in the personal computer 20 and function in a known manner. For the sake of brevity, other components, such as a joystick, sound board and speakers are not illustrated in FIG. 1.

The personal computer 20 may also include a network interface 39 to permit operation in a networked environment using logical connections to one or more remote computers, such as a remote computer 40. The remote computer 40 may be another personal computer, a server, a router, a network PC, a peer device or other common network node, and typically includes many or all of the elements described above relative to the personal computer 20, although only a memory storage device 42 has been illustrated in FIG. 1. The logical connections depicted in FIG. 1 include a local area network (LAN) 43 and a wide area network (WAN) 44. Such networking environments are commonplace in offices, enterprise-wide computer networks, intranets and the Internet.

When used in a LAN networking environment, the personal computer 20 is connected to the LAN 43 through the network interface 39. When used in a WAN networking environment, the personal computer 20 typically includes a modem 45 or other means for establishing communications over the wide area network 44, such as the Internet. The modem 45, which may be internal or external, permits communication with remote computers 46-50. In a networked environment, program modules depicted relative to the personal computer 20, or portions thereof, may be stored in the remote memory storage device 42 via the LAN 51 or stored in a remote memory storage device 52 via the WAN 44. It will be appreciated that the network connections shown are exemplary and other means of establishing a communications link between the computers may be used.

The present invention is embodied in a system 100 illustrated in the functional diagram of FIG. 2. In a typical telephone communication, an originating telephone 102 is operated by the caller to place a call to a destination telephone 104. The originating telephone generates signals that are detected by a central office switch 106 operated by a local exchange carrier (LEC) 108. The LEC 108 is the telephone service provider for the calling party. The originating telephone 102 is coupled to the central office switch 106 via a communication link 110. As those skilled in the art can appreciate, the communication link 110 may be a hard-wired connection, such as a fiber optic, copper wire, or the like.

Alternatively, the communication link 110 may be a wireless communication link if the originating phone 102 is a cellular telephone or some other form of wireless telephone.

Similarly, the destination telephone 104 is coupled to a central office switch 116 operated by a local exchange carrier (LEC) 118. The destination telephone 104 is coupled to the central office switch 116 via a communication link 120. The communication link 120 may be a hard-wired communication link or a wireless communication link, as described above with respect to the communication link 110. The present invention is not limited by the specific form of communication link or central office switch.

The LEC 108 establishes a communication link with the LEC 118. As illustrated in FIG. 2, the communication link between the LEC 108 and the LEC 118 is through a long distance carrier (LDC) 124. The LEC 108 establishes a communication link 126 with the LDC 124 which, in turn, establishes a communication link 128 with the LEC 118. If

the telephone call from the originating telephone **102** to the destination telephone **104** is not a long distance call, the LDC **124** is not required. In this case, the communication link **126** may couple the LEC **108** directly to the LEC **118**. The use of the system **100** with other telephone configurations are illustrated in other figures.

To place a telephone call, the caller activates the originating telephone **102** to dial in the telephone number corresponding to the destination telephone number **104**, thereby establishing the communication link **110** with the central office switch **106**. In turn, the central office switch **106** establishes the communication link **126** (via the LDC **124**, if necessary), thus establishing a communication link with the central office switch **116**. In a conventional telephone system, the central office switch **116** establishes the communication link **120** to the destination telephone **104** causing the destination telephone to ring. If the callee picks up the destination telephone, a complete communication link between the originating telephone **102** and the destination telephone **104** has been established. This is sometimes referred to as "terminating" the telephone call. The specific telecommunications protocol used to establish a telephone communication link between the originating telephone **102** and the destination telephone **104** is well known in the art and need not be described herein. The preceding description of techniques used to establish the telephone communication link are provided only as a basis for describing the additional activities performed by the system **100**.

With the system **100**, the central office switch **116** does not initially establish the telephone communication link **120** with the destination telephone **104** to cause the telephone to ring. Instead, the central office switch **116** establishes a communication link **132** with a computer network **134**, such as the Internet. As those skilled in the art can appreciate, the Internet is a vast multi-computer network coupled together by data links having various communication speeds. Although the Internet **134** may use a variety of different communication protocols, a well-known communication protocol used by the Internet is a Transmission Control Protocol/Internet Protocol (TCP/IP). The transmission of data on the Internet **134** using the TCP/IP is known to those skilled in the art and need not be described in greater detail herein.

The central office switch **116** utilizes conventional telephone communication protocols, which may be different from the TCP/IP communication protocols used by the Internet **134**. The system **100** includes a communication interface **136** to translate data between the two communication protocols. The communication interface **136** includes a telephone interface portion **138** and an Internet interface portion **140**. The telephone interface portion **138** is coupled to the central office switch **116** via the communication link **132** such that communications occurring on the communication link **132** utilize the telephone communication protocol. The Internet interface portion **140** communicates via the Internet using conventional communication protocols, such as TCP/IP.

The communication interface **136** may be implemented on a computing platform that functions as a server. The conventional components of the computing platform, such as a CPU, memory, and the like are known to those skilled in the art and need not be described in greater detail herein. The telephone interface portion **138** may comprise an Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI) to communicate with the central office switch **116**. The ISDN PRI, which may be implemented on a plug-in computer card, provides information to the telephone interface portion **138**, such as automatic number identification (ANI), dialed number identification service (DNIS), and the like. As is known, ANI provides the telephone number of the caller's telephone (e.g., the originating telephone **102**) while the DNIS allows the number the caller dialed (e.g., the destination telephone **104**) to be forwarded to a computer system. These data may be considered "keys" which may be used by the system **100** to identify the caller and the callee. Thus, the central office switch **116** provides information which may be used to access the affiliation list **150** for the destination telephone **104**.

The Internet interface portion **140** may be conveniently implemented with a computer network card mounted in the same computing platform that includes the ISDN PRI card. However, it is not necessary for satisfactory operation of the system **100** that the interface cards be co-located in the same computing platform. It is only required that the telephone interface portion **138** communicate with the Internet interface portion **140**. The Internet interface portion **140** receives the incoming data (e.g., the ANI, DNIS, and the like) and generates Internet compatible commands. The specific form of the Internet commands using, by way of example, TCP/IP, are within the scope of knowledge of one skilled in the art and need not be described herein. As will be described below, data provided by the central office switch **116** will be used to access data on the Internet and use that data to determine the manner in which a telephone call will be processed.

The Internet **134** stores an affiliation list **150**, which may be established by the user of the destination telephone **104**. Data stored within the affiliation list **150** is accessed by the central office switch **116** to determine the manner in which the call from the originating telephone **102** will be processed. Details of the affiliation list **150** are provided below. The Internet **134** also includes an Internet controller **152** which communicates with a callee computer **154** via a network link **156**. The communication between the callee computer **154** and the Internet **134** is a conventional communication link used by millions of computers throughout the world. For example, the callee computer **154** may be a personal computer (PC) containing a communication interface, such as a modem (not shown). The network link **156** may be a simple telephone communication link using the modem to communicate with the Internet **134**. The Internet controller **152** functions in a conventional manner to communicate with the callee computer **154** via the network link **156**. Although the communication link **132** and the network link **156** are both communication links to the Internet, the network link **156** is a conventional computer connection established over a telephone line, a network connection, such as an Ethernet link, or the like. This conventional network link **156** is significantly different from the communication link **132** between the central office switch **116** and the Internet **134**. The central office switch **116** establishes the communication link **132** to access data on the Internet and uses that accessed data to determine how to process an incoming call for the destination telephone **104**. The network link **156** is a computer-to-computer connection that may simply use a telephone as the physical layer to establish the network link.

In the system **100**, the central office switch **116** receives an incoming call from the originating telephone **102** via the central office switch **106** and, optionally, the LDC **124**. Rather than immediately establishing the communication link **120** and generating a ring signal at the destination telephone **104**, the central office switch **116** establishes the

communication link 132 and communicates with the Internet 134 via the communication interface 136. The purpose of such communication is to access the affiliation list 150 and thereby determine the manner in which the user of the destination telephone 104 wishes calls to be processed.

FIG. 3 illustrates the system 100 for a telephone system configuration in which the originating telephone 102 and the destination telephone 104 are both serviced by the same local exchange carrier 108. The originating telephone 102 establishes the communication link 110 with the central office switch 106 in the manner described above. The central office switch 106 establishes the communication link 126 directly with the central office switch 116 without the need for the IXC 124 (see FIG. 2). The central office switch 116 operates in the manner described above. That is, the central office switch 116 does not immediately establish the communication link 120, but does establish the communication link 132 with the Internet 134. For the sake of simplicity, FIG. 3 does not illustrate the communication interface 136. However, those skilled in the art will appreciate that the central office switch 116 accesses the affiliation list 150 via the communication interface 136 (see FIG. 2).

For the sake of simplicity, FIG. 3 also does not show the Internet controller 152 and the callee computer 154. However, those skilled in the art can appreciate that those portions of the system may also be present in the embodiment illustrated in FIG. 3. However, it should be noted that the callee computer 154 and the Internet controller 152 need only be used to edit the affiliation list 150. The call processing by the central office switch 116 does not depend on the presence of the Internet controller 152 or the callee computer 154. That is, the central office switch 116 accesses the affiliation list 150 via the communication interface 136 regardless of the presence of the callee computer 154.

In yet another telephone system configuration, illustrated in FIG. 4, the originating telephone 102 and the destination telephone 104 are not only serviced by the same local exchange carrier 108, but are connected to the same central office switch 116. However, the fundamental operation of the system 100 remains identical to that described above with respect to accessing the affiliation list 150. That is, the originating telephone 102 establishes the communication link 110 with the central office switch 116. However, the central office switch 106 need not establish the communication link 126 with any other central office switch since the destination telephone 104 is also connected to that same central office switch.

In this telephone system configuration, the central office switch 116 accesses the affiliation list 150 on the Internet 134 via the communication link 132 (see FIG. 2) in the manner described above. For the sake of simplicity, FIG. 4 does not illustrate the communication interface 136. However, those skilled in the art will recognize that the communication interface 136 operates to convert communication signals between telephone protocol used by the central office switch 106 and the Internet communication protocol used by the Internet 134. In addition, FIG. 4 also does not illustrate the Internet controller 152 and the callee computer 154. As noted above with respect to FIG. 3, the Internet controller 152 and callee computer 154 are not necessary for proper operation of the system 100. The callee computer 154 is typically used in the system 100 to edit the affiliation list 150.

The affiliation list 150 is illustrated in greater detail in the functional block diagram of FIG. 5. The affiliation list comprises a series of sublists, illustrated in FIG. 3 as a forward list 160, a reverse list 162, a block list 164, and an allow list 166. The forward list 160 contains a list of Internet subscribers whose Internet activity a user wishes to monitor. This list is sometimes referred to as a "buddy" list. When the user operates the callee computer 154 on the Internet 134, the Internet controller 152 accesses the forward list 160 via an affiliation list input/output (I/O) interface 170 to determine which Internet subscribers contained within the forward list are currently active on the Internet 134. In conventional Internet operation, the Internet controller 152 sends a message to the callee computer 154 indicating which Internet subscribers on the forward list 160 are currently active on the Internet 134.

The forward list 160 is a list of Internet subscribers whose activity is reported to the user. Other Internet subscribers may have their own forward list (not shown) and may monitor the Internet activity of the user. When the user accesses the Internet 134 with the callee computer 154, that activity can be monitored by others. With the system 100, it is possible to determine who is monitoring the user's Internet activity. The reverse list 162 contains a list of Internet subscribers who have placed the user in their forward list. That is, the reverse list 162 contains a list of Internet subscribers who have placed the user in their buddy list. With the reverse list 162, the user can determine who is monitoring his Internet activity.

The block list 164 contains a list of Internet subscribers that the user does not want to monitor his Internet activity. That is, the user's Internet activity will not be provided to any Internet subscriber contained in the block list 164. Thus, even if a particular Internet subscriber has placed the user on their forward list, the presence of that particular Internet subscriber's name on the block list 164 will prevent the user's Internet activity from being reported to the particular Internet subscriber. The use of the block list 164 provides certain security assurances to the user that their Internet activity is not being monitored by any undesirable Internet subscribers.

The allow list 166 contains a list of Internet subscribers for whom the user may wish to communicate with but whose Internet activity the user does not wish to monitor.

The system 100 combines the capabilities of the affiliation list 150 with telephone switching technology to filter incoming calls to the destination telephone 104. For example, the user may specify that only calls from Internet subscribers contained in the forward list 154 may contact the user via the destination telephone 104. Alternatively, the user may specify that a calling party whose name is contained in the forward list 160 or the allow list 166 may place a call to the destination telephone 104. As will be discussed in greater detail below, the system 100 allows the user to create general conditional processing, such as blocking calls or allowing calls. However, the user can also create specific conditional processing for individual callers or based on the user's current status or preferences.

The central office switch 116 accesses the affiliation list 150 via the communication link 132 and determines whether the calling party is in a list (e.g., the forward list 160) that the user wishes to communicate with. If the calling party is contained within an "approved" list, the central office switch 116 establishes the communication link 120 and sends a ring signal to the destination telephone 104. Thus, the user can pick up the telephone with the knowledge that the calling party is an individual with whom the user wishes to communicate.

Conversely, if the calling party is not contained within an approved list, such as the forward list 160 or the allow list

166, the central office switch 116 will not establish the communication link 120 with the destination telephone 104. Thus, the user will not be bothered by undesirable phone calls. In one embodiment, the central switch office simply will not establish the communication link 120 and the calling party will recognize that the call did not go through. Alternatively, the central office switch 116 may generate a signal indicating that the destination telephone 104 is busy. In this alternative embodiment, the calling party will receive a busy signal on the originating telephone 102. Thus, the user has the ability to filter incoming calls by creating a list of those individuals with whom the user wishes to communicate.

It should be noted that the affiliation list 150 may be dynamically altered by the user to add or delete individuals, change individuals from one list to another, or to change the call processing options for a particular list depending on the user's preferences. For example, the user may want to accept all calls from any source at certain times of the day. Under these circumstances, the user can edit the allow list 166 to accept calls from any calling party. Alternatively, the user may still maintain the block list 164 such that calls will not be processed from certain specified parties even if the user is willing to accept calls from any other source. Under other circumstances, the user may not wish to communicate with any individuals. In this instance, the user may indicate that all calling parties are on the block list 164. Thus, the central office switch 116 will access the Internet 134 in real-time and review data in the affiliation list 150 to thereby process incoming calls for the user in accordance with the rules present in the affiliation list.

The discussion above provides examples of the central office switch 116 processing calls from a calling party in accordance with their presence or absence of certain lists in the affiliation list 150. For example, a call from a party on the forward list 160 will be connected to the destination telephone 104 (see FIG. 2) while a call from a party on the block list 164 will not be put through to the destination telephone. However, the system 100 also allows the selection of call processing options on an individual basis rather than simply on the presence or absence in a particular list. For example, the user can edit the allow list 166 to specify that certain individuals are "allowed" while other individuals may be allowed, conditionally allowed, or blocked all together. If the individual calling party has an associated status indicating that they are allowed, the central office switch 116 will process the incoming call and connect it to the destination telephone 104. If the individual calling party has an associated blocked status, the central office switch 116 will not process the call and will not connect it to the destination telephone 104.

Furthermore, the user may attach conditional status to individual callers or to calling lists. Conditional status may be based on factors, such as the time of day, current availability of the user, work status, or the like. For example, the user may accept calls from certain work parties during specified periods of the day (e.g., 9:00 a.m.–11:00 a.m.), block calls from selected calling parties during other periods of time (e.g., 12:00–1:00 p.m.), or allow calls during a business meeting only from certain calling parties (e.g., the boss). These conditional status criteria may be applied to individuals or to one or more lists in the affiliation list 150.

FIG. 6 illustrates sample data entries in the allow list 166. The allow list 166 may include data, such as a name, Internet subscriber name, and one or more phone numbers associated with the individual data entry. It should be noted that the calling party need not have an Internet subscriber name for proper operation of the system 100. That is, the central office switch 116 accesses the allow list 166 utilizing the calling party number and need not rely on any email addresses or other Internet subscriber identification for proper operation. The allow list 166 may also include an email alias in addition to or in place of the Internet subscriber name. Some Internet subscribers prefer to "chat" with other subscribers utilizing an alias rather than their actual Internet subscriber name. The data of FIG. 6 illustrates one possible embodiment for the allow list 166. However, those skilled in the art can appreciate that the allow list 166 may typically be a part of a large database (not shown). Database operation is well known in the art, and need not be described in greater detail herein. The database or other form of the forward list 160 may be satisfactorily implemented using any known data structure for storage of data. For example, the various lists (e.g., the allow list 166, the reverse list 162, the block list 164 and the allow list 166) may all be integrated within a single database structure. The present invention is not limited by the specific structure of the affiliation list 150 nor by the form or format of data contained therein.

Rather than incoming call filtering on the basis of presence in a particular list, such as the allow list 166, as illustrated in FIG. 6, the affiliation list 150 may contain status data on an individual basis. In this event, the central office switch 116 (see FIG. 2) processes the incoming call in accordance with the designated status for that individual. In the example illustrated in FIG. 7, the affiliation list 150 contains one individual with an "allowed" status, one individual with a "blocked" status, and one individual with a "conditional" status based on user-selected criteria. In the example of FIG. 7, the user-selected criteria may be based on the particular phone from which the call is originating as well as the time of day in which the call is originated. For example, the user may wish to allow all calls from a particular number, such as an caller's work number. However, calls from another number, such as the caller's home phone, may be blocked. Other calls, such as from a caller's cellular telephone, may be allowed only at certain times of day. FIG. 7 is intended to illustrate some of the call processing options that are available to the user. As can be appreciated, a variety of different conditional status criteria may be applied to one or more potential calling parties. However, a common feature of the system 100 is that the telecommunication system. (e.g., the central office switch 116) determines calling pat status on the basis of information stored on the Internet and processes the incoming call in accordance with the user-specified criteria. Moreover, the system 100 operates in real-time to process the incoming call in accordance with the user-specified criteria.

The Internet 134 may be conveniently used as a storage area for the caller specified criteria. The advantage of such data storage on the Internet is that the data is widely accessible to the user. This provides a convenient mechanism for entering new caller data or editing existing caller data. The user can access the affiliation list 150 with the callee computer 154 via the network link 156. In contrast, the central office switch 116 may access the affiliation list 150 via the communication link 132, which may typically be a high-speed communication link. In addition, FIGS. 2, 4, and 5 illustrate the central office switch 116 as the telecommunication component that accesses the Internet 134. It is convenient for operational efficiency to have the central office switch (e.g., the central office switch 116) to which the destination telephone 104 is connected perform such Internet access. It is at this stage of the telephone call processing that the telecommunication system may most conveniently

determine the user-specified caller status. However, those skilled in the art will recognize that the status check may be performed by other portions of the telecommunication system, such as the central office switch 106, the LDC 124, or the like. Thus, the present invention is not limited by the particular telecommunication component that establishes the communication link with a network which the user-specified caller status data is stored.

In addition, the system 100 can be readily implemented as an "add-on" component of the telecommunication system and need not be integrated with the central office switch 116. For example, the conventional central office switch provides the ability to divert calls based on certain call conditions, such as "Call Forward No Answer," which may be used to divert an incoming call to voicemail or "Call Forward Busy," which may also divert the incoming call to voicemail. To implement the system 100 with an add-on processor, the system may optionally include a Switch to Computer Applications Interface (SCAI) 174 and a call processor 176. The dashed lines of FIG. 4 are intended to illustrate an alternative configuration of the system 100. This alternative configuration can also be implemented with other telephone system configurations, such as illustrated in FIGS. 2 and 3. The SCAI 174 is a telecommunication protocol that allows switches to communicate with external computers. Data, such as caller and callee telephone numbers, and status information, such as Call Forward Busy, are provided to the SCAM 174 by the central office switch 116.

The call processor 176 performs the functions described above to process the call in accordance with the user-specified criteria. That is, the call processor 176 receives caller and callee data from the SCAI 174 and accesses the affiliation list 150 via the communication interface 136 (see FIG. 2). The call processor 176 uses user-specified call processing criteria to generate instructions for the central office switch 116. The instructions are provided to the central office switch 116 via the SCAI 174. Those skilled in the art will appreciate that the SCAI 174 is but one example of the Open Application Interface (OAI) that can be used with the central office switch 116.

As noted above, the system 100 can process a call intended for the destination telephone 104, block a call, or generate a busy signal at the originating telephone 102. However, the system 100 also operates with voicemail and permits a number of different customized outgoing messages. FIG. 4 illustrates a voicemail system 180 having a storage area containing one or more outgoing messages 182. For example, the voicemail system 180 can play an outgoing message 182 informing the caller that "the party you are calling only accepts calls from designated callers. Please leave a message." If calls are blocked only at certain times, the outgoing message 182 can say "the party you are calling does not accept calls between 11:30 a.m. and 1:00 p.m. Please leave a message or call back after 1:00 p.m." The outgoing message can also reflect callee availability by playing a message such as "The party you are calling is in a meeting. Please leave a message or call back in X minutes" where X reflects the amount of time before the meeting is expected to end. That information can be manually provided to the affiliation list 150 by the user or automatically derived from a computerized scheduling program on, by way of example, the callee computer 154 (see FIG. 2).

Computerized scheduling programs, such as Microsoft® D Schedule Plus, can be used on the callee computer 154 (see FIG. 2). It is known that such scheduling programs can be accessed via a computer network or downloaded to a hand-held computing device to track appointments. The system 100 can access such computerized scheduling programs and download appointments and scheduled meetings into the affiliation list 150. The outgoing messages 182 can be automatically selected on the basis of the user's computerized schedule. Thus, the system 100 permits the user to schedule his day (e.g., meetings, lunch time, in office/available for calls, in office/unavailable for calls, etc.) on a computerized scheduling program and to process calls in accordance with the computerized schedule and even select outgoing messages automatically based on the user's schedule.

The operation of the system 100 is illustrated in the flowchart of FIG. 7. At a start 200, the calling party has placed a call from the originating telephone 102 (see FIG. 2) to the destination telephone 104. In step 202, the central office switch 116 has received call data from the originating telephone 102. The received call data includes the destination telephone number of the destination telephone 104 and identification data indicating the originating telephone 102 as the source of the present call. Use of automatic number identification (ANI) is a well-known technique for providing identification data indicating the originating telephone 102 as the source of the present call. While the specific implementation of ANI data, sometimes referred to as caller ID, may not be uniformly implemented throughout the United States, the ANI data is typically delivered between the first and second rings. In the present invention, the central office switch 116 (see FIG. 2) does not initiate a ring signal to the destination telephone 104 until after determining the status of the calling party based on the ANI. In future implementations, telecommunication companies may transmit other forms of caller identification, such as caller name, Internet address, email alias, or the like. The system 100 operates satisfactorily with any form of caller identification. The only requirement for the system 100 is that some form of caller identification be provided. The call is processed in accordance with the user-specified criteria in the affiliation list 150 for the identified caller.

In step 204, the central office switch 116 (see FIG. 2) establishes the communication link 132 with the Internet 134. Although step 204 illustrates the system 100 as actively establishing the communication link 132 with the Internet 134, those skilled in the art will recognize that the system 100 can utilize a continuous high-speed data link between the central office switch and the Internet. Thus, it is not necessary to establish a network link for each and every incoming call processed by the central office switch 116. As previously described, the communication interface 136 translates data between the telephone protocol and the Internet protocol. In step 206, the system 100 accesses the affiliation list 150 for the user (i.e., the called party). In an exemplary embodiment, the telephone number of the destination telephone 104 or other callee identification is used as an index or pointer to a specific location within the database where the affiliation list 150 for the particular user may be found. Database operation in general, and techniques for locating specific items within a database in particular are known to those skilled in the art and need not be described herein.

In decision 210, the system 100 determines whether the caller identification data is on the forward list 160 (see FIG. 3). If the caller identification data is present in the forward list, the result of the decision 210 is YES. In that event, the system 100 proceeds to FIG. 6B where the call is processed in accordance with the rules associated with the forward list 160.

If the caller identification data is not present in the forward list 160 (see FIG. 3), the result of decision 210 is

NO. In that event, the system 100 moves to decision 212 to determine whether the caller identification data is in the allow list 166. If the caller identification data is present in the allow list 166, the result of decision 214 is YES. In that event, the system 100 proceeds to decision 216 where the call is processed in accordance with the rules associated with the allow list 166. If the caller identification data is not present in the allow list 166, the result of decision 216 is NO.

In decision 218, the system 100 determines whether the caller identification data is present in the reverse list 162. If the caller identification data is present in the reverse list 162, the system 100 proceeds to the step 220 where the call is processed in accordance with the rules associated with the reverse list 162. If the caller identification data is not present in the reverse list, the result of decision 218 is NO. In that event, the system moves to decision 216 to determine whether the caller is present on the block list 164. If the caller is present on the block list 164, the result of decision 222 is YES. In that event, the system proceeds to step 224 where the call is processed in accordance with the rules associated with the block list. If the caller identification data is not present in the block list 164, the result of decision 222 is NO. This indicates that the caller identification data is not present in any of the user-specified lists in the affiliation list 150. In that event, the system moves to step 226 where the call may be processed in accordance with user-specified rules of processing anonymous or unidentified calls. The flowchart of FIG. 8 illustrates the operation of the system 100 with multiple lists wherein the call processing rules are designated for each list. In this embodiment, the call is processed on the basis of the presence or absence of the caller identification data in a particular list. However, as previously discussed, the affiliation list 150 (see FIG. 5B) may include user-specified status criteria for individual callers. In this embodiment, the system 100 processes the call on the basis of the user-specified status criteria associated with the individual caller rather than on the basis of the caller's presence or absence in a specific list. In that event, the system 100 may simply access the user affiliation list (see step 206 in FIG. 7) and process the call in accordance with the user-specified status criteria for the individual caller. If the caller identification data is not present in the affiliation list 160, the call may be processed using user-specified call processing criteria for unidentified callers, as shown in step 226.

Thus, the system 100 allows the user to specify call processing rules for a plurality of different caller lists or for individual callers within a list. The caller lists may be readily edited in accordance with the changing desires of the user. The user may alter the call processing rules in accordance with various times of day, work conditions, or even the personal mood of the user. For example, the user may process all calls during certain times of the day, such as when the user is at work. However, when the user arrives home, subsequent calls may be processed in accordance with a different set of rules, such as accepting no calls during dinner time or after a certain time at night.

These rules may be applied differentially to different ones of the list in the affiliation list 150. For example, the user may accept calls from any calling party on the forward list 160 (see FIG. 3) or the allow list 166 during the evening hours. However, after a certain time at night, the caller may accept calls only from calling parties on the forward list 160. Thus, the system 100 allows great flexibility in the user selection of calling rules and lists. The system 100 allows the user to filter incoming calls in accordance with generalized rules or in accordance with highly specific rules.

In addition to filtering incoming calls to the destination telephone 104, the system 100 can monitor the status or activity of both the caller and the callee and establish a communication link between the originating telephone 102 and the destination telephone 104 when the status data indicates that both the caller and callee are available for a telephone conversation. The system 100 has been previously described with respect to callee status monitoring and processing of incoming calls in accordance with the user-selected (i.e., the callee-selected) call processing criteria. Similar status monitoring can be performed for the caller. As illustrated in FIG. 9, the system 100 may include a caller computer 184, which is coupled to the Internet via the communication link 132. For the sake of clarity, FIG. 9 illustrates the callee computer 154 and the caller computer 184 as connected to the Internet 134 through a single Internet controller 152. However, those skilled in the art will appreciate that the Internet 134, or any computer network, includes many network controllers that function as a gateway to the network. Thus, the system 100 typically includes a large number of Internet controllers 152.

In addition, for the sake of clarity, Figure illustrates only a single affiliation list 150. However, those skilled in the art will appreciate that separate affiliation lists exist for the originating telephone 102 and the destination telephone 104. The central office switch 116 (or the call processor 176) access the appropriate affiliation list via the network connection 132 and apply the appropriate call processing rules for each telephone.

FIG. 9 also illustrates a keyboard 154a and mouse 154b coupled to the callee computer 154 for use in a conventional fashion. Similarly, the caller computer 184 includes a keyboard 184a and a mouse 184b. The computer operating system, such as the Windows® operating system, is capable of monitoring user activity on the computer. For example, the operating system on the callee computer 154 can detect user activity on the keyboard 154a or the mouse 154b. By monitoring this activity, the operating system can determine the user's status and activate certain software programs, such as a screen saver, when no user activity has been detected for a certain period of time. Under these circumstances, the operating system may determine that the callee computer 154 has entered an "idle" state. Similarly, operating system on the caller computer 184 may perform similar functions to determine user activity on the caller computer. Using the principles of the present invention, the callee computer 154 and the caller computer 184 may report the current status to the affiliation list 150 for each respective computer.

The system 100 can monitor computer activity and generate signals to both the originating telephone 102 and the destination telephone 104 when the callee computer 154 and the caller computer 184 are not in the idle state. The fact that both computers' are not in the idle state indicates that the users of each respective computer may be available for a telephone conversation. In addition, the system 100 can apply call processing rules that may also govern operation of the telephone portion of the system 100. For example, the callee computer 154 may be in an "active" state (as opposed to the idle state) but the user has indicated that he should not be disturbed at the present time. Thus, the central office switch 116 or the call processor 176 accesses the affiliation list 150 for the destination telephone 104 to determine the callee-selected call processing criteria. In addition, the central office switch 116 or the call processor 176 can access the affiliation list 150 for the caller and apply any caller-selected call processing rules. For example, the caller computer 184

may be in the active state, but the caller status in the affiliation list 150 may indicate that the caller is in a meeting and is, therefore, unavailable for a telephone call with the callee. In this manner, the system 100 can monitor computer activity and determine when the caller and callee may both be available for a telephone call and further applies call processing criteria for both the caller and callee. The call processing criteria for the caller and callee as well as the current status of the callee computer 154 and the caller computer 184 are stored within the respective affiliation lists 150 on the Internet 134. This data may be accessed by the central office switch 116 or the call processor 176 via the network connection 132 in the manner previously described.

In operation, the system allows a caller to indicate a desire to establish a telephone communication link with a specified callee. The caller can use the originating telephone 102 or the caller computer 184 to initiate the call processing by the system 100. The system 100 monitors the caller and callee activities and call processing rules and, when appropriate for both parties, establishes a telephone communication link by sending signals from the central office switch 116 to the originating telephone to generate a ring signal. The central office switch 116 also generates appropriate signals to generate ring signal at the destination telephone 104.

As can be appreciated, the originating telephone 102 communicates with the central office switch 116 using the communication link 110 while the caller computer 184 communicates with the Internet 134 using the communication link 132. The communication link 132 may be a second telephone line, a network connection, such as an Ethernet connection, or the like. If the user has two telephone lines, the telephone number of the telephone (e.g., the destination telephone 104) can be different from the telephone number associated with the computer (e.g., the callee computer 154). However, the system 100 must be aware of an association between the telephone and the computer. This is particularly important if the status of the computer (i.e., idle or active) is used as one of the call processing criteria. The system 100 can monitor the activity of a computer (e.g., the callee computer 154) in order to establish a telephone communication link with an associated telephone (e.g., the destination telephone 104). It is of no value to monitor a user's computer status at one location and call a completely unrelated telephone at a different location. For example, it is of no value to monitor the callee's computer at work and then to call the callee's home telephone number.

In other implementations, such as with a home computer, only a single telephone line may serve the function of both the communication link 110 and the communication link 132. Under these circumstances, the caller may use the caller computer 184 to indicate a desire to establish the telephone communication link and then must terminate the communication link 132 so that the central office switch may generate the appropriate signals on the communication link 110 at a point in time when the callee call processing criteria and the caller call processing criteria are both met. It should be further noted that this implementation will preclude the use of the status (i.e., idle or active) of the caller computer 184 since the communication link 132 is not active.

Similarly, the destination telephone 104 and the callee computer 154 may be connected to the central office switch 116 and the Internet 134 via separate communication links (i.e., the communication link 120 and the communication link 132, respectively). However, the system 100 may also be implemented with a single phone line. The callee may use the callee computer 154 and the communication link 132 to generate or edit the callee call processing criteria in the affiliation list 150. However, the user must then terminate the communication link 132 to permit the central office switch 116 to establish the communication link 120. As noted above, a single phone line precludes the use of computer status monitoring (i.e., idle or active) for the callee computer 154 since the status cannot be monitored via the communication link 132.

The operation of the system 100 to establish a communication link with both the originating telephone 102 and the destination telephone 104 is illustrated in the flowchart of FIG. 10 where, at a start 250, it is assumed that the caller and callee both have data in their respective affiliation lists. As previously noted, the affiliation list 150 for each individual may comprise separate sublists, such as illustrated in FIG. 5, or a single data structure containing call processing criteria, such as allowing or blocking individual calls (see FIG. 7) or establishing conditional criteria, such as time restrictions, current user status (e.g., in a meeting), or the current status of the user's computer (e.g., the idle or active status of the callee computer 154). Furthermore, as previously noted, user status can be automatically provided to the affiliation list 150 by a computerized schedule program.

In step 252, the caller indicates a desire to establish a telephone communication link with the callee. In a conventional communication system, the caller picks up the originating telephone and dials the telephone number for the destination telephone 104. However, in accordance with this aspect of the system 100, the caller may indicate the desire to establish a telecommunication link using the caller computer 184 and placing the callee telephone number (i.e., the telephone number of the destination telephone 104) on a call list, such as the forward list 160 (see FIG. 5). By placing the callee on the forward list, the system 100 can access the callee affiliation list to determine whether the callee computer 154 is active on the Internet.

With the callee telephone number (i.e., the telephone number of the destination telephone 102) placed on the call list, the system 100 can determine the call processing criteria of both the caller and the callee, and process the request for a telephone call in accordance with those rules. In step 254, the system 100 establishes a communication link with the Internet 134. As previously noted, the central office switch 116 may directly establish the communication link 132 with the Internet 134 or may use the SCAI 174 and call processor 176 to communicate with the Internet. It should be noted that the telephone portion of the system may have a continuous data link with the Internet via the central office switch 116 or the call processor 176. Thus, it is not necessary to continuously establish and tear down the communication link 132.

In step 258, the system 100 accesses the callee affiliation list 150. In step 260, the system 100 accesses the caller affiliation list 150. As previously noted, the physical location of each affiliation list in unimportant to the satisfactory operation of the system. The only requirement is that the affiliation list is accessible via the computer network, such as the Internet 134.

In decision 262, the system 100 applies the callee call processing criteria and determines whether the present calling conditions meet the callee criteria. This includes testing whether the caller is contained within one of the sublists illustrated in FIG. 5 or if the status associated with the call origination data indicates that the caller is allowed or blocked, or the like. If the present calling conditions do not meet the callee criteria, the result of decision 262 is NO. In that event, the system 100 can return to step 258 to again

access the callee affiliation list. As those skilled in the art can appreciate, the callee affiliation list may be updated by the callee (typically via the callee computer 154) which may change the result of decision 262.

If the current call does meet the callee call processing criteria, the result of decision 262 is YES. In that event, the system 100 uses the data from the caller affiliation list 150 to determine whether the present call meets the caller call processing criteria. Although the caller indicated a desire to establish a telephone link with the callee, the caller may not be available for an immediate phone call. For example, the caller may have a meeting scheduled to begin, but expects to be available for a phone call following the meeting. The caller can manually set the call processing criteria, such as indicating the desired time of the telephone call. Alternatively, the caller call processing criteria may be automatically supplied to the caller affiliation list 150 through the use of a computerized scheduling program or the like. The system 100 may also monitor the status of the caller computer 184 to determine caller availability. For example, the caller may indicate an availability for a phone call after a predetermined time. The system 100 can detect the change in the state of the caller computer 184 from the idle state to the active state and interpret that as an indication that the caller is now available for a telephone call. The system can apply these conditions individually or in various combinations to determine the availability of the caller and callee. If the call does not meet the caller call processing criteria, the result of decision 264 is NO. In that event, the system 100 can return to step 258 to access the affiliation lists for the callee and caller, respectively, and thus continuously monitor the callee and caller call processing criteria to determine an appropriate time to make a phone call.

If the call does meet the caller call processing criteria, the result of decision 264 is YES. In that event, in step 266 the system 100 causes the central office switch 116 to send the appropriate ring signals to the originating telephone 102 and ring signals to the destination telephone 104. In this manner, the telephone system follows the call processing guidelines of both caller and callee stored on a computer network to control the processing of the call on the telephone network.

Although the example illustrated in FIG. 10 illustrates a continuous process of checking call processing criteria against the current call conditions, those skilled in the art appreciate that other possible actions can be taken by the system 100. For example, the caller may be on the block list 164 (see FIG. 5). In this condition, the call will never meet the callee call processing criteria. The system 100 thus will never establish a communication link. The system 100 can send a message to the caller computer 184 indicating that the callee does not accept calls in this manner and to leave a message on the voicemail system 180. Alternatively, the system 100 can establish a telephone communication link to the originating telephone 102 and provide a similar message. As discussed above with respect to FIG. 4, a variety of voice mail messages can be provided to the user. The system 100 may establish a telephone communication link to the originating telephone 102 and play the appropriate outgoing message 182 (see FIG. 4). As noted above, the system 100 can apply call processing rules derived from any source, such as the current status (e.g., idle or active) of the callee computer 154 or the caller computer 184, the presence or absence on one of the sublists in FIG. 5 (e.g., the block list 164), the status of one party (e.g., the allowed status of the caller), callee or caller status data provided by computerized scheduling systems, or the like. The system 100 advantageously allows multiple forms of call processing criteria to be stored in the network, such as the Internet 134, and accessed by the telephone system, such as the central office switch 116 or the call processor 176. Those skilled in the art will also recognize that the embodiment of the system 100 shown in FIG. 9 can be implemented with various telephone system configurations, such as those illustrated in FIGS. 2 and 3, or any other telephone system configuration. Furthermore, the system 100 is not limited by the specific component of the telephone system that establishes the network link 132 with the affiliation list 150. Although FIG. 9 illustrates the central office switch 116 or the call processor 176 as the component that establishes the network link, those skilled in the art will recognize that other components, such as the central office switch 106 (see FIG. 2), the LDC 124, or the like can establish the network link 132. Thus, the system 100 is not limited by the specific component of the telephone communication system that establishes the network link 132.

From the foregoing it will be appreciated that, although specific embodiments of the invention have been described herein for purposes of illustration, various modifications may be made without deviating from the spirit and scope of the invention. For example, the system discussed herein uses, by way of example, the Internet 134 to store the affiliation list 150. However, the system 100 can be implemented with other computer networks or as a portion of a telephone switch, such as the central office switch 116. The telephone service provider can provide a customer with an affiliation list and some means to control the list as a value-added telephone service. The central office switch 116 accesses the internal affiliation list and processes the incoming calls in accordance with the user-specified criteria contained therein. Accordingly, the invention is not limited except as by the appended claims.

What is claimed is:

1. In a system that includes a telephone network and a computer network with one or more users, wherein each user is connected through a user computer the computer network and is logically connected through the computer network to the telephone network, a method of determining when to establish telephone communication between two parties, at least one of whom is a user connected to said computer network, comprising:

at the computer network, receiving information from the telephone network that a first party from whom a call is originating desires to establish telephone communication with a second party;

at the computer network, monitoring activity of a user computer connected to the computer network and associated with the second party;

at the computer network, storing a set of pre-determined rules for determining when the second party is available to take a call from the first party;

at the computer network, using the set of a pre-determined rules to process i) the information received from the telephone network regarding the call being originated by the first party, and ii) information regarding the monitored activity of the user computer of the second party, to determine when the second party is available to take the call originated by the first party; and

using the information processed at the computer network to facilitate connecting the call originated by the first party through the telephone network to the second party.

2. A method as recited in claim 1, further comprising, at the computer network, monitor activity of a user computer

connected to the computer network and associated with the first party, wherein using the set of pre-determined rules is also performed using information regarding the monitored activity of the user computer of the first party.

3. A method as recited in claim 1, wherein using the information processed at the computer network to facilitate connecting the call comprises sending control signals to the telephone network to cause the telephone network to connect the call.

4. A method as recited in claim 1, wherein the predetermined rules are associated with an affiliation list of the second party and wherein the first party is referenced by the buddy list.

5. A method as recited in claim 1, wherein monitoring activity of a user computer connected to the computer network and associated with the second party comprises monitoring activity of an input device of the user computer.

6. A method as recited in claim 1, wherein the pre-defined rules specify whether the second party accepts telephone calls from the first party.

7. In a system that includes a telephone network and a computer network with one or more users, and wherein each user is connected through a user computer to the computer network and is logically connected through the computer network to the telephone network, a computer program product comprising:

a computer readable medium for carrying computer executable instructions for implementing at the computer network a method of determining when to establish telephone communication between two parties, at least one of whom is a user connected to said computer network, and wherein said method comprises:

at the computer network, receiving information from the telephone network that a first party from whom a call is originating desires to establish telephone communication with a second party;

at the computer network, monitoring activity of a user computer connected to the computer network and associated with the second party;

at the computer network, storing a set of predetermined rules for determining when the second party is available to take a call from the first party; and

at the computer network, using the set of predetermined rules to process i) the information received from the telephone network regarding the call being originated by the first party, and ii) information regarding the monitored activity of the user computer of the second party, to determine when the second party is available to take the call originated by the first party.

8. A computer program product as recited in claim 7, wherein the method further comprises using the information processed at the computer network to facilitate connecting the call originated by the first party through the telephone network to the second party.

9. A computer program product as recited in claim 7, wherein the pre-determined rules specify whether the second party accepts telephone calls from the first party.

10. A computer program product as recited in claim 7, wherein the pre-determined rules define how the telephone call is to be processed based on the time of the day of the telephone call.

11. A computer program product as recited in claim 7, wherein the method further comprises, at the computer network, monitoring activity of a user computer connected to the computer network and associated with the first party, wherein using the set of pre-determined rules is also performed using information regarding the monitored activity of the user computer of the first party.

12. In a system that includes a telephone network and a computer network with one or more users, and wherein each user is connected through a user computer to the computer network and is logically connected through the computer network to the telephone network, a method of determining when to establish telephone communication between two parties, each of whom is a user connected to said computer network, comprising:

at the computer network, monitoring activity of the user computers associated with both a first and a second party;

at the computer network, receiving information from the telephone network that the first party is originating a call to the second party;

at the computer network, storing a set of pre-determined rules for determining when the second party is available to take a call from the first party;

at the computer network, using the set of pre-determined rules to process i) the information received from the telephone network regarding the call being originated by the first party, and ii) information regarding the monitored activity of the user computers of the first and second parties, to determine when the second party is available to take the call originated by the first party; and

using the information processed at the computer network to facilitate connecting the call originated by the first party through the telephone network to the second party.

13. A method as recited in claim 12, wherein using the information processed at the computer network to facilitate connecting the call comprises sending control signals to the telephone network to cause the telephone network to connect the call.

14. A method as recited in claim 12, wherein the predetermined rules are associated with an affiliation list of the second party and wherein the first party is referenced by the buddy list.

15. A method as recited in claim 12, wherein monitoring activity of a user computer connected to the computer network and associated with the second party comprises monitoring activity of an input device of the user computer associated with the second party.

16. A method as recited in claim 12, wherein the predefined rules specify whether the second party accepts telephone calls from the first party.

17. In a system that includes a telephone network and a computer network with one or more users, and wherein each user is connected through a user computer to the computer network and is logically connected through the computer network to the telephone network, a computer program product comprising:

a computer readable medium for carrying computer executable instructions for implementing at the computer network a method of determining when to establish telephone communication between two parties, each of whom is a user connected to said computer network, wherein said method comprises:

at the computer network, monitoring activity of the user computers associated with both the first and second parties;

at the computer network, receiving information from the telephone network that the first party is originating a call to the second party;

at the computer network, storing a set of predetermined rules for determining when the second party is available to take a call from the first party; and

at the computer network, using the set of pre-determined rules to process i) the information received from the telephone network regarding the call being originated by the first party, and ii) information regarding the monitored activity of the user computers of the first and second parties, to determine when the second party is available to take the call originated by the first party.

18. A computer program product as recited in claim **17**, wherein the method further comprises using the information processed at the computer network to facilitate connecting

the call originated by the first party through the telephone network to the second party.

19. A computer program product as recited in claim **17**, wherein the pre-determined rules specify whether the second party accepts telephone calls from the first party.

20. A computer program product as recited in claim **17**, wherein the pre-determined rules define how the telephone call is to be processed based on the time of the day of the telephone call.

* * * * *

EXHIBITS 2-9

REDACTED IN THEIR ENTIRETY

EXHIBIT 10

US005999965A

United States Patent [19]

Kelly

[11] **Patent Number: 5,999,965**

[45] **Date of Patent: Dec. 7, 1999**

[54] **AUTOMATIC CALL DISTRIBUTION SERVER FOR COMPUTER TELEPHONY COMMUNICATIONS**

[75] Inventor: **Keith C. Kelly**, Deerfield Beach, Fla.

[73] Assignee: **NetSpeak Corporation**, Boca Raton, Fla.

[21] Appl. No.: **08/914,714**

[22] Filed: **Aug. 19, 1997**

Related U.S. Application Data

[60] Provisional application No. 60/024,234, Aug. 20, 1996.

[51] **Int. Cl.**^{6} **G06F 13/00**

[52] **U.S. Cl.** **709/202**

[58] **Field of Search** .... 364/DIG. 1 MS File, 364/DIG. 2 MS File; 709/200, 201, 202, 218, 219, 220, 226, 229, 231, 232, 238

[56] **References Cited**

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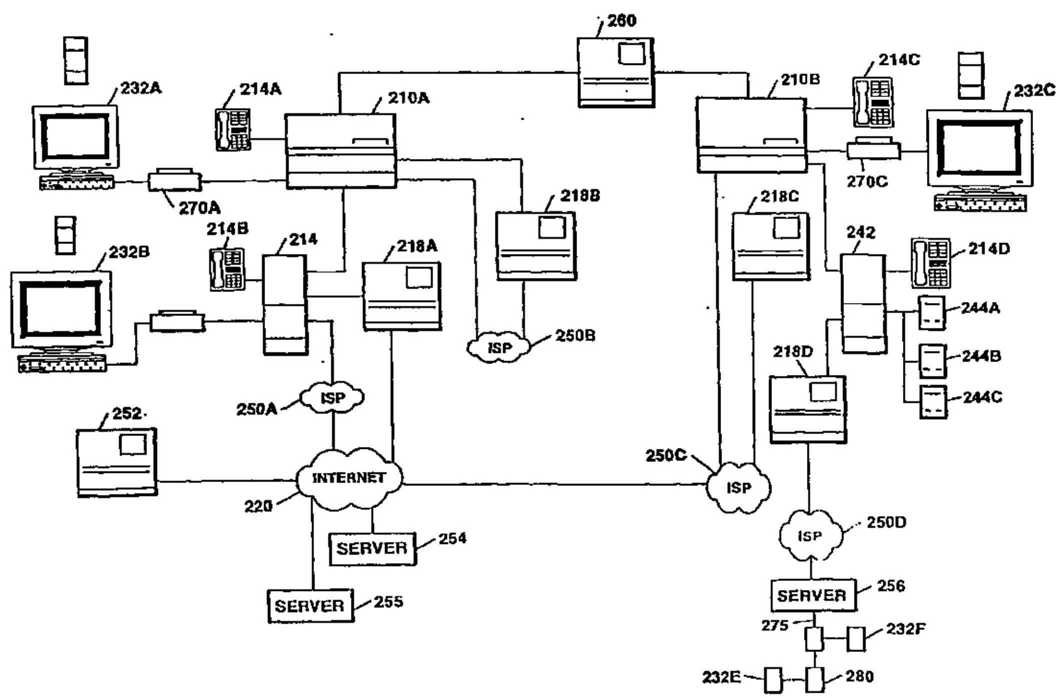
Primary Examiner—Robert B. Harrell

Attorney, Agent, or Firm—Kudirka & Jobse, LLP

[57] **ABSTRACT**

An automatic call distribution system capable of receiving incoming calls originating on either circuit-switched or packet-switched networks utilizes an automatic call distribution (ACD) server for receiving and routing incoming calls and a control center module for dynamically configuring a plurality of agent processes to which the incoming calls may be transferred. The agent processes, control center and ACD server may be separated geographically, but operatively coupled via a computer network. The incoming calls contain user information which enables calls to be routed by the ACD server according to a plurality of different criteria. A graphic user interface enables a system user to dynamically monitor the status of agent processes and reconfigure both queues and the agent processes associated with a queue in response to call loads and agent resource availability.

3 Claims, 14 Drawing Sheets



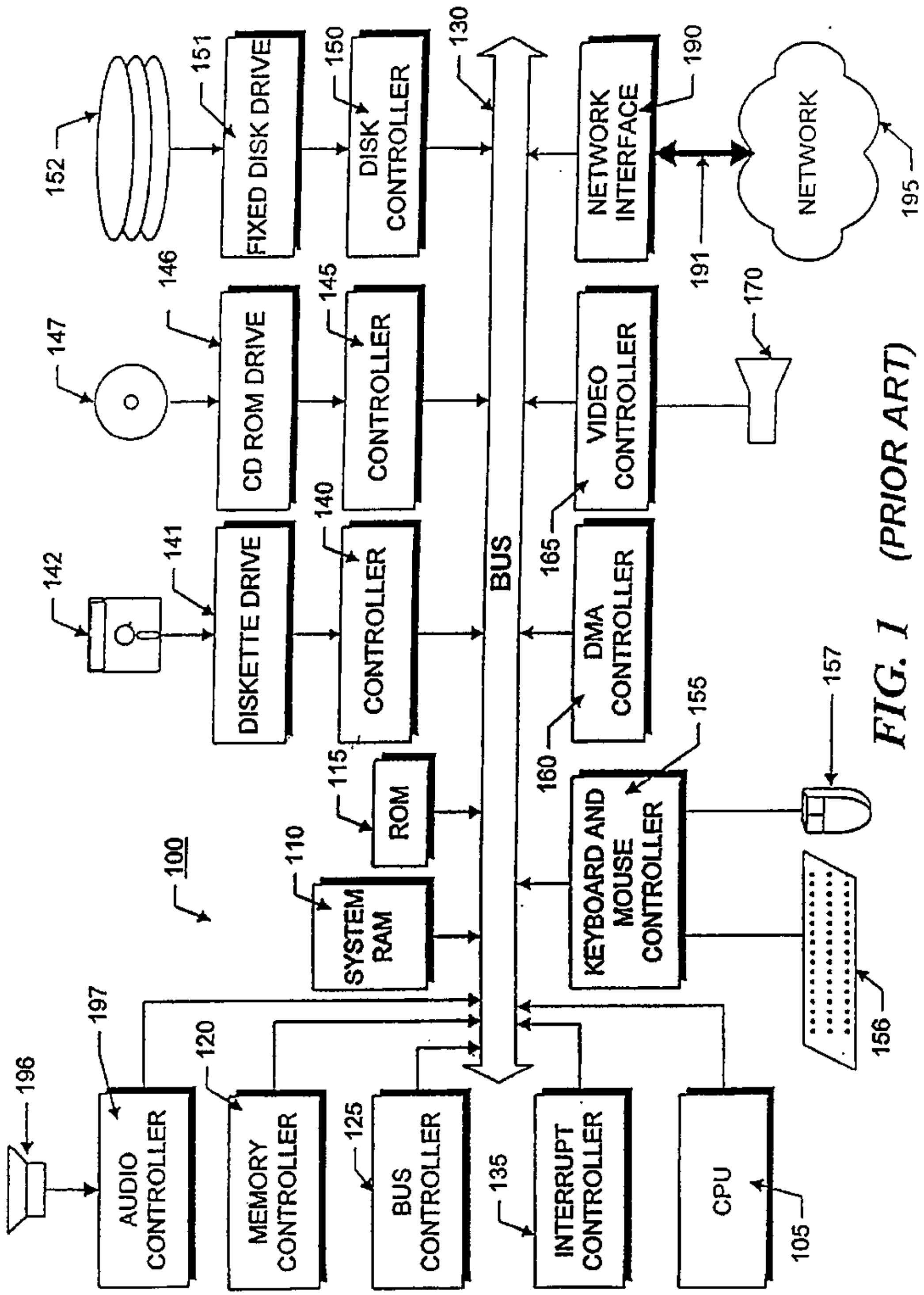


FIG. 1 (PRIOR ART)

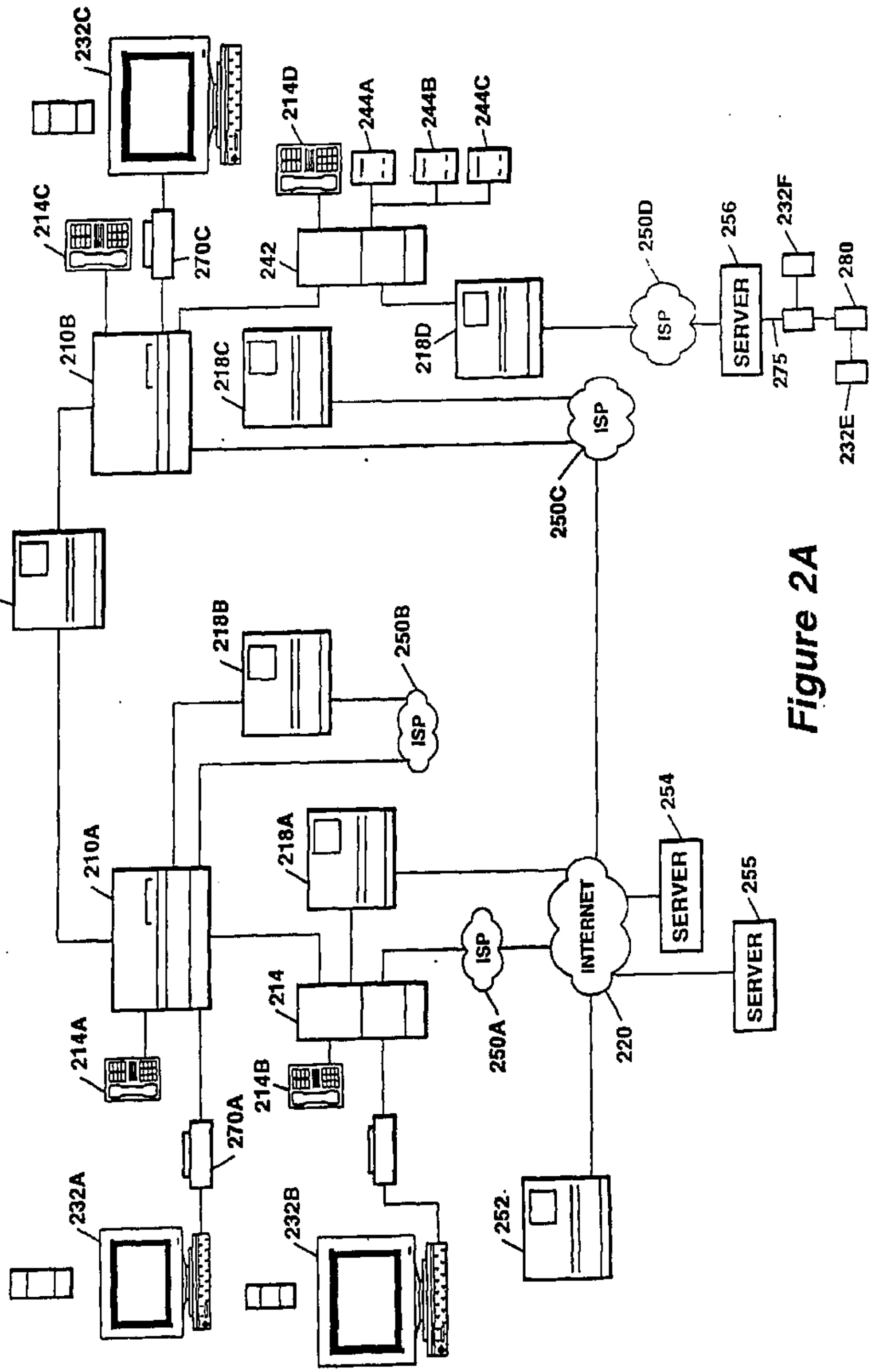

232C
214C
270C
210B
214D
244A
244B
244C
242
218C
218D
250D
ISP
256
SERVER
232F
280
275
232E
ISP
250C
260
218B
250B
ISP
210A
218A
254
SERVER
255
SERVER
INTERNET
220
ISP
250A
214
214A
214B
270A
232A
232B
252

Figure 2A

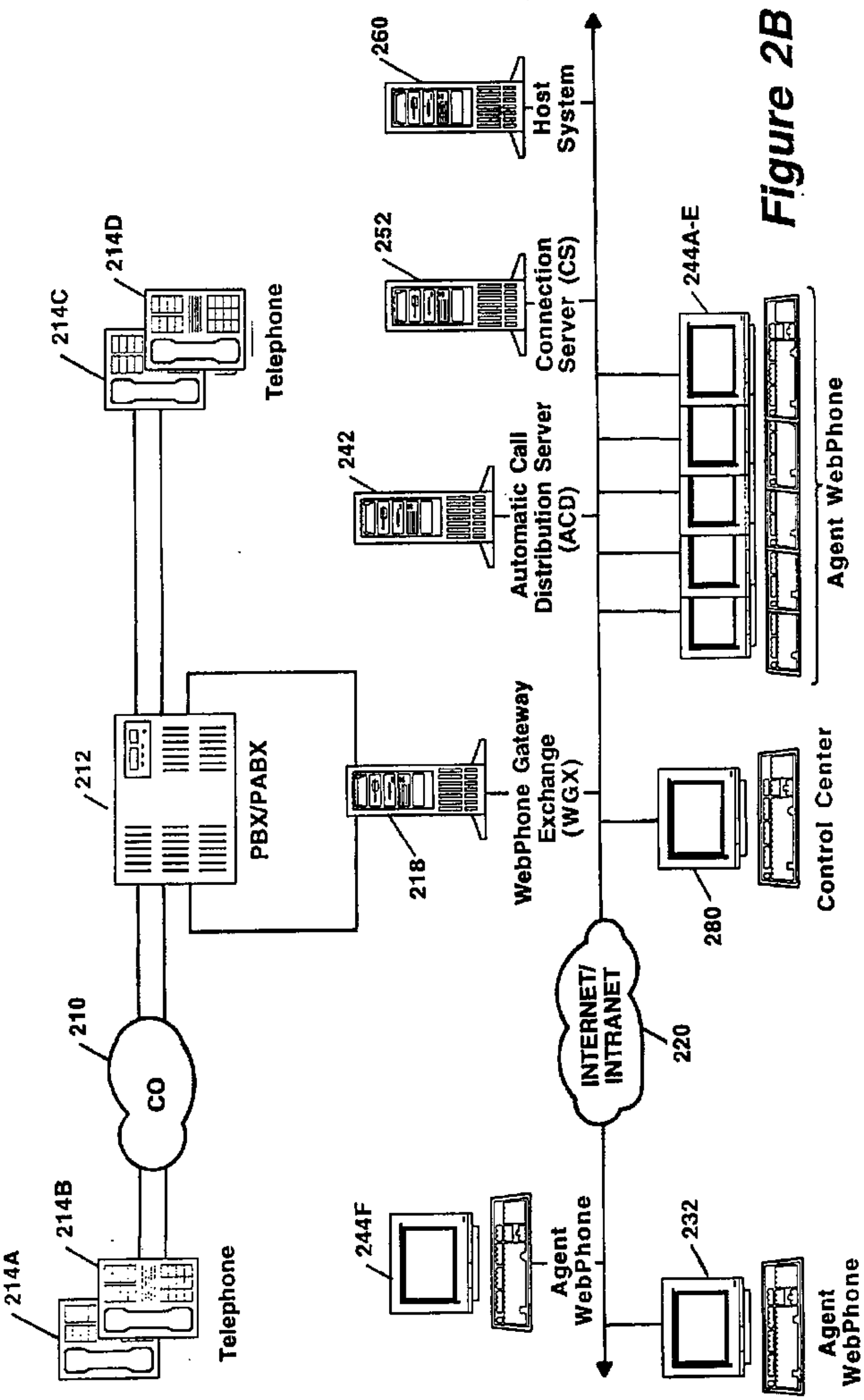

214A
214B
Telephone
210
CO
212
PBX/PABX
214C
214D
Telephone
218
WebPhone Gateway Exchange (WGX)
242
Automatic Call Distribution Server (ACD)
252
Connection Server (CS)
260
Host System
244F
Agent WebPhone
232
Agent WebPhone
INTERNET/ INTRANET
220
280
Control Center
244A-E
Agent WebPhone

Figure 2B

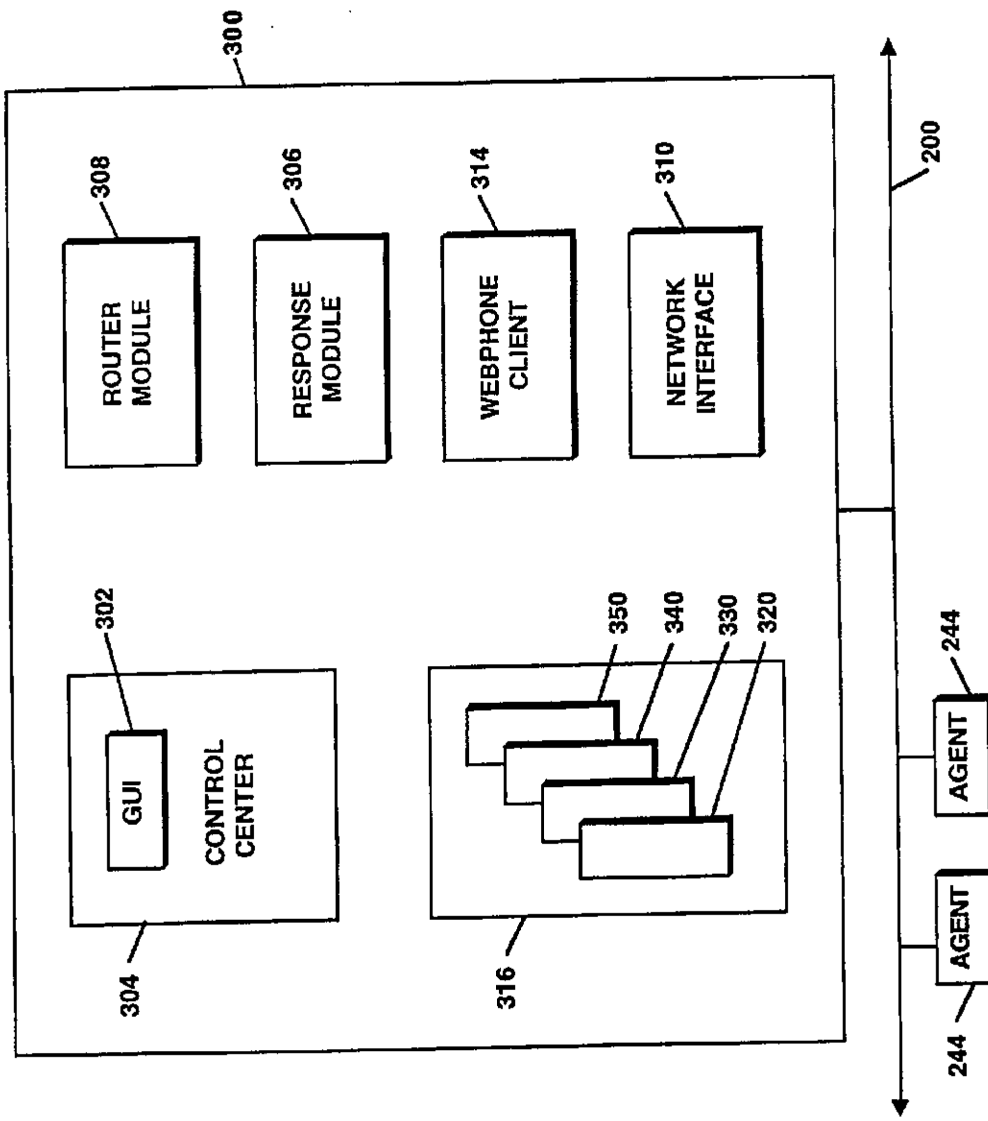

300
308
ROUTER MODULE
306
RESPONSE MODULE
314
WEBPHONE CLIENT
310
NETWORK INTERFACE
200
302
GUI
CONTROL CENTER
304
350
340
330
320
316
244
AGENT
AGENT
244

Figure 3A

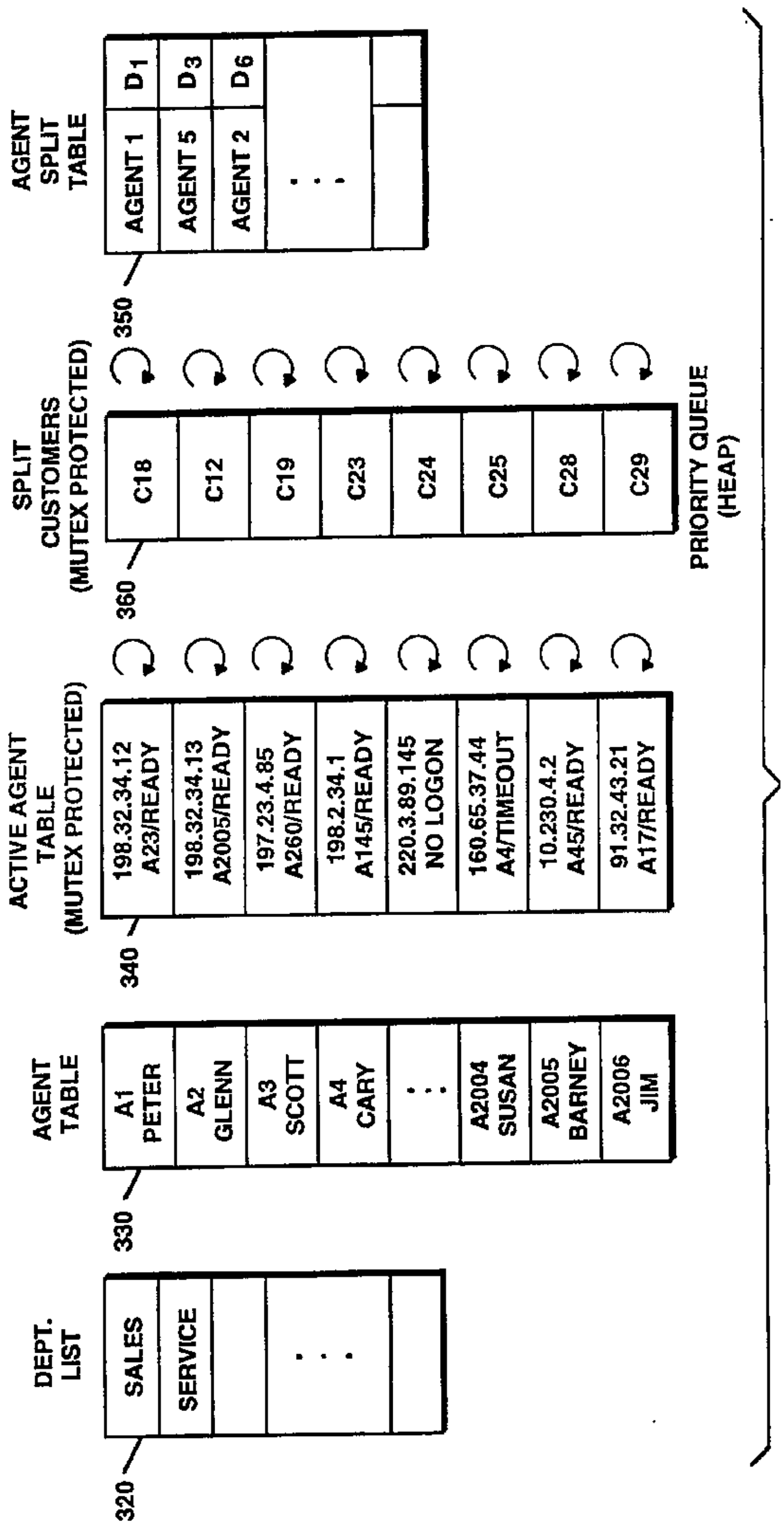

DEPT. LIST
SALES
SERVICE
. . .
320
AGENT TABLE
A1 PETER
A2 GLENN
A3 SCOTT
A4 CARY
. . .
A2004 SUSAN
A2005 BARNEY
A2006 JIM
330
ACTIVE AGENT TABLE (MUTEX PROTECTED)
198.32.34.12 A23/READY
198.32.34.13 A2005/READY
197.23.4.85 A260/READY
198.2.34.1 A145/READY
220.3.89.145 NO LOGON
160.65.37.44 A4/TIMEOUT
10.230.4.2 A45/READY
91.32.43.21 A17/READY
340
SPLIT CUSTOMERS (MUTEX PROTECTED)
C18
C12
C19
C23
C24
C25
C28
C29
PRIORITY QUEUE (HEAP)
360
AGENT SPLIT TABLE
AGENT 1 D1
AGENT 5 D3
AGENT 2 D6
. . .
350

Figure 3B

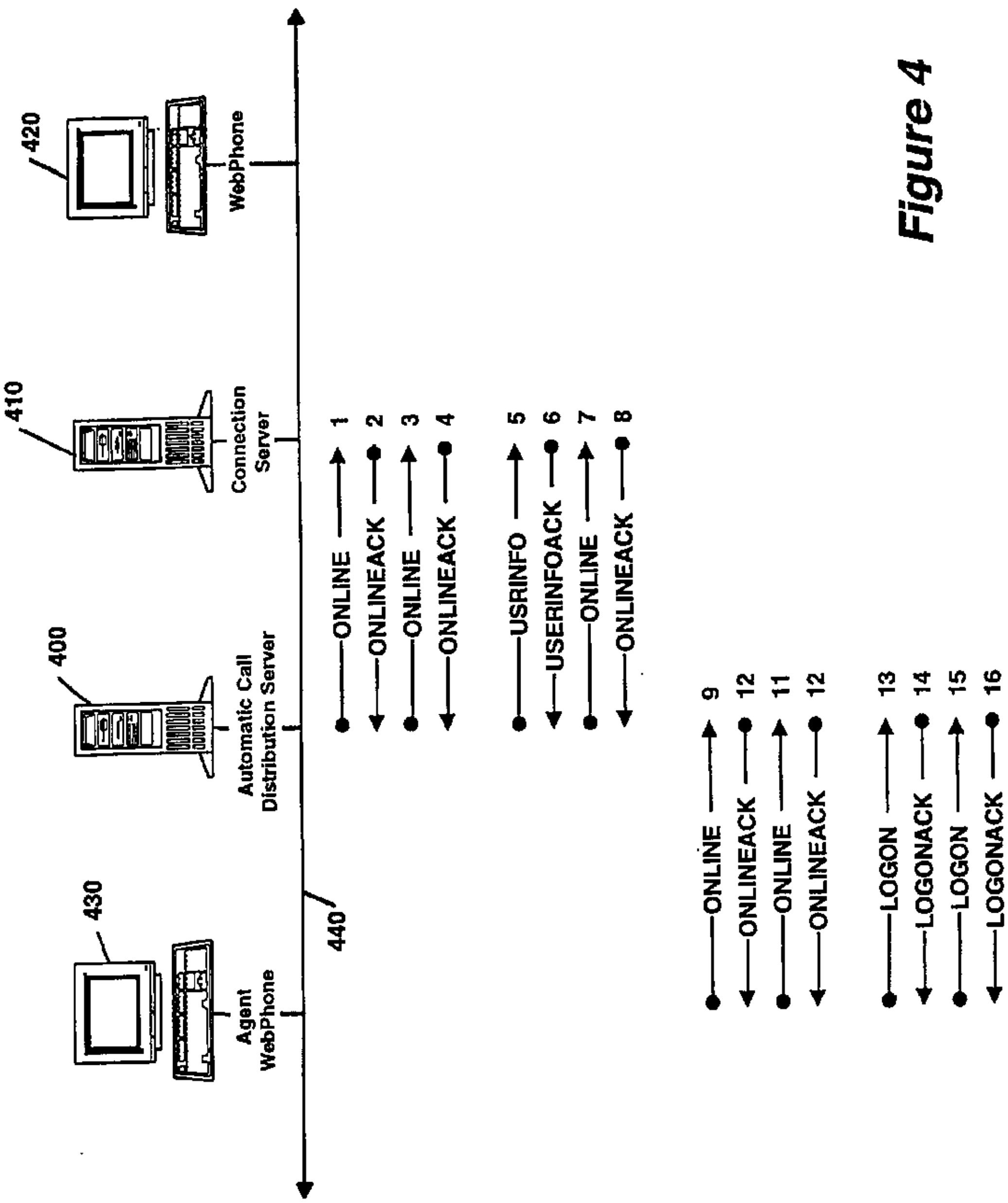

420
WebPhone
410
Connection Server
400
Automatic Call Distribution Server
430
Agent WebPhone
440
ONLINE 1
ONLINEACK 2
ONLINE 3
ONLINEACK 4
USRINFO 5
USERINFOACK 6
ONLINE 7
ONLINEACK 8
ONLINE 9
ONLINEACK 12
ONLINE 11
ONLINEACK 12
LOGON 13
LOGONACK 14
LOGON 15
LOGONACK 16

Figure 4

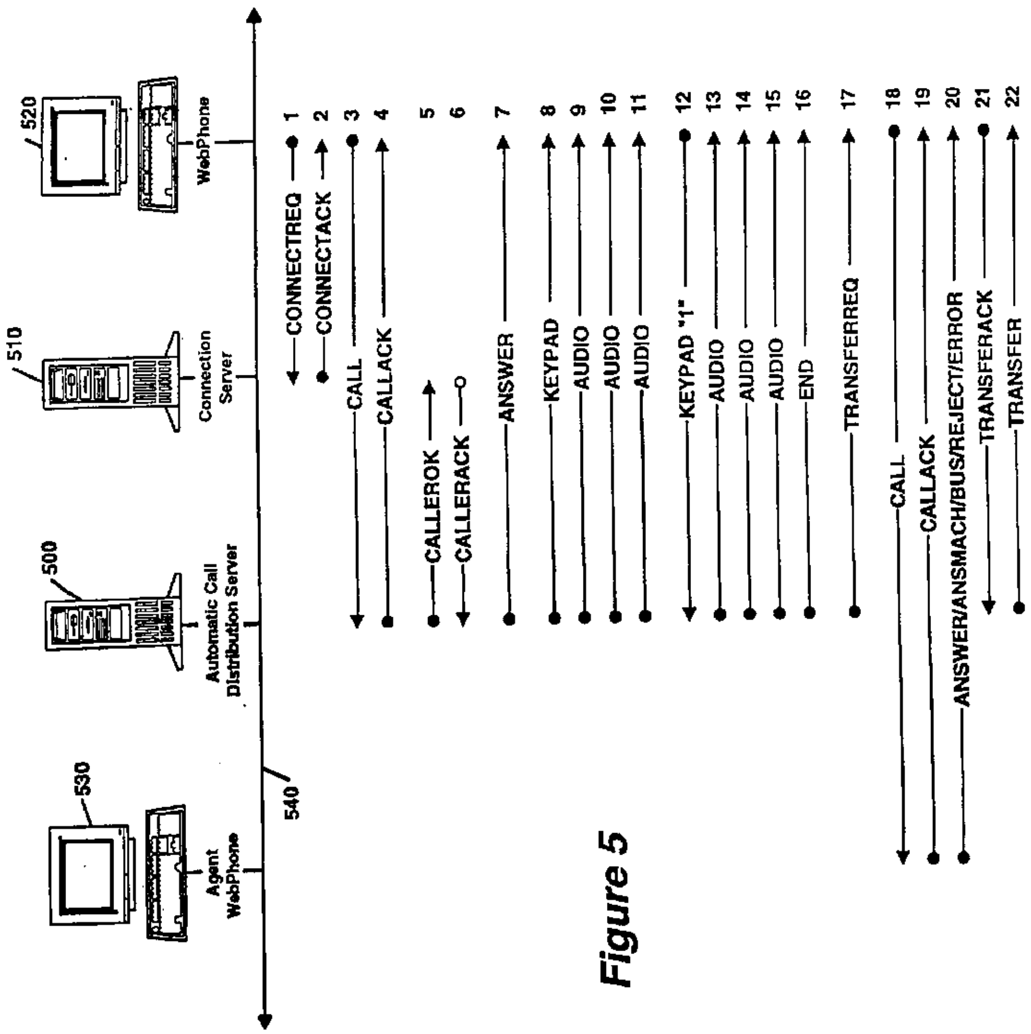

520
WebPhone
510
Connection Server
500
Automatic Call Distribution Server
530
Agent WebPhone
540
CONNECTREQ
CONNECTACK
CALL
CALLACK
CALLEROK
CALLERACK
ANSWER
KEYPAD
AUDIO
AUDIO
AUDIO
KEYPAD "1"
AUDIO
AUDIO
AUDIO
END
TRANSFERREQ
CALL
CALLACK
ANSWER/ANSMACH/BUS/REJECT/ERROR
TRANSFERACK
TRANSFER
1
2
3
4
5
6
7
8
9
10
11
12
13
14
15
16
17
18
19
20
21
22

Figure 5

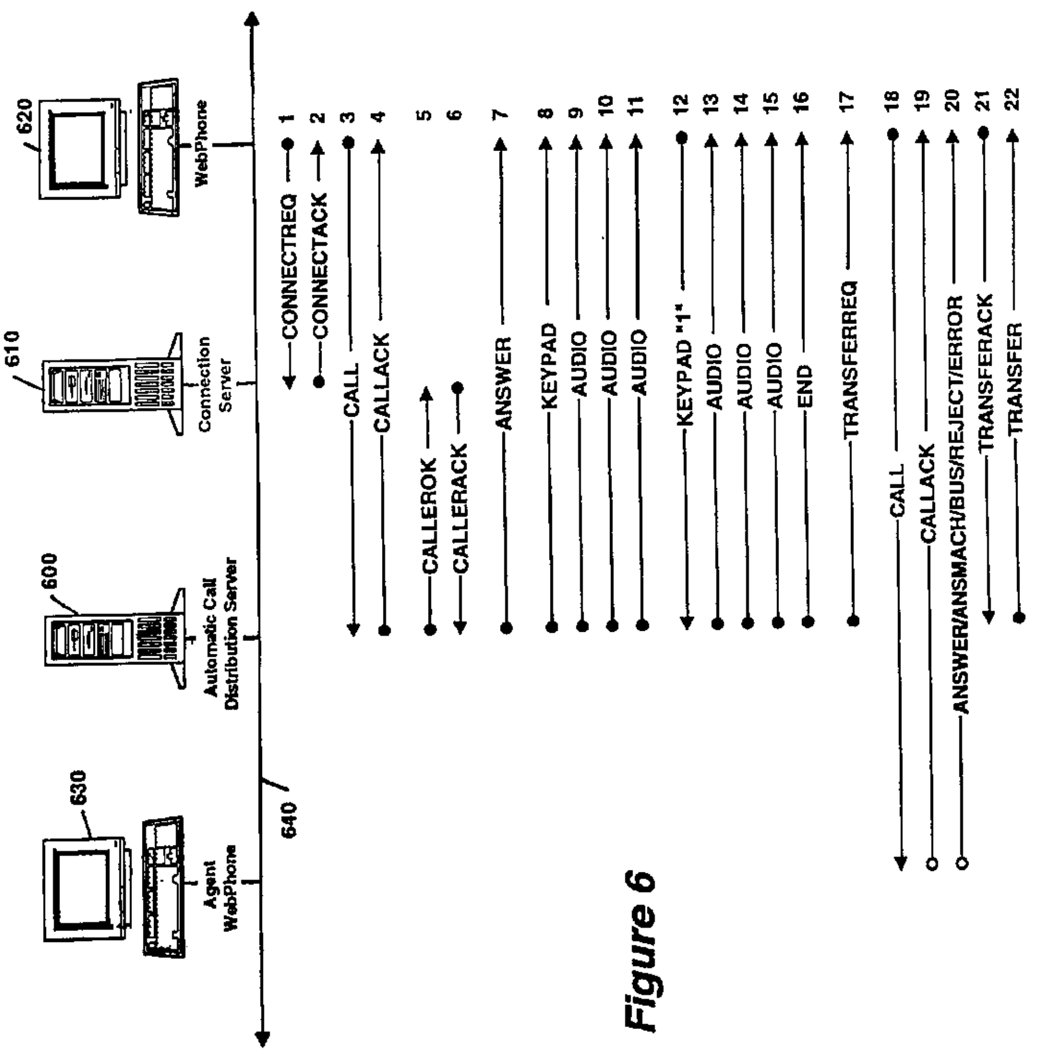

620
WebPhone
610
Connection Server
600
Automatic Call Distribution Server
630
Agent WebPhone
640
CONNECTREQ 1
CONNECTACK 2
CALL 3
CALLACK 4
CALLEROK 5
CALLERACK 6
ANSWER 7
KEYPAD 8
AUDIO 9
AUDIO 10
AUDIO 11
KEYPAD "1" 12
AUDIO 13
AUDIO 14
AUDIO 15
END 16
TRANSFERREQ 17
CALL 18
CALLACK 19
ANSWER/ANSMACH/BUS/REJECT/ERROR 20
TRANSFERACK 21
TRANSFER 22

Figure 6

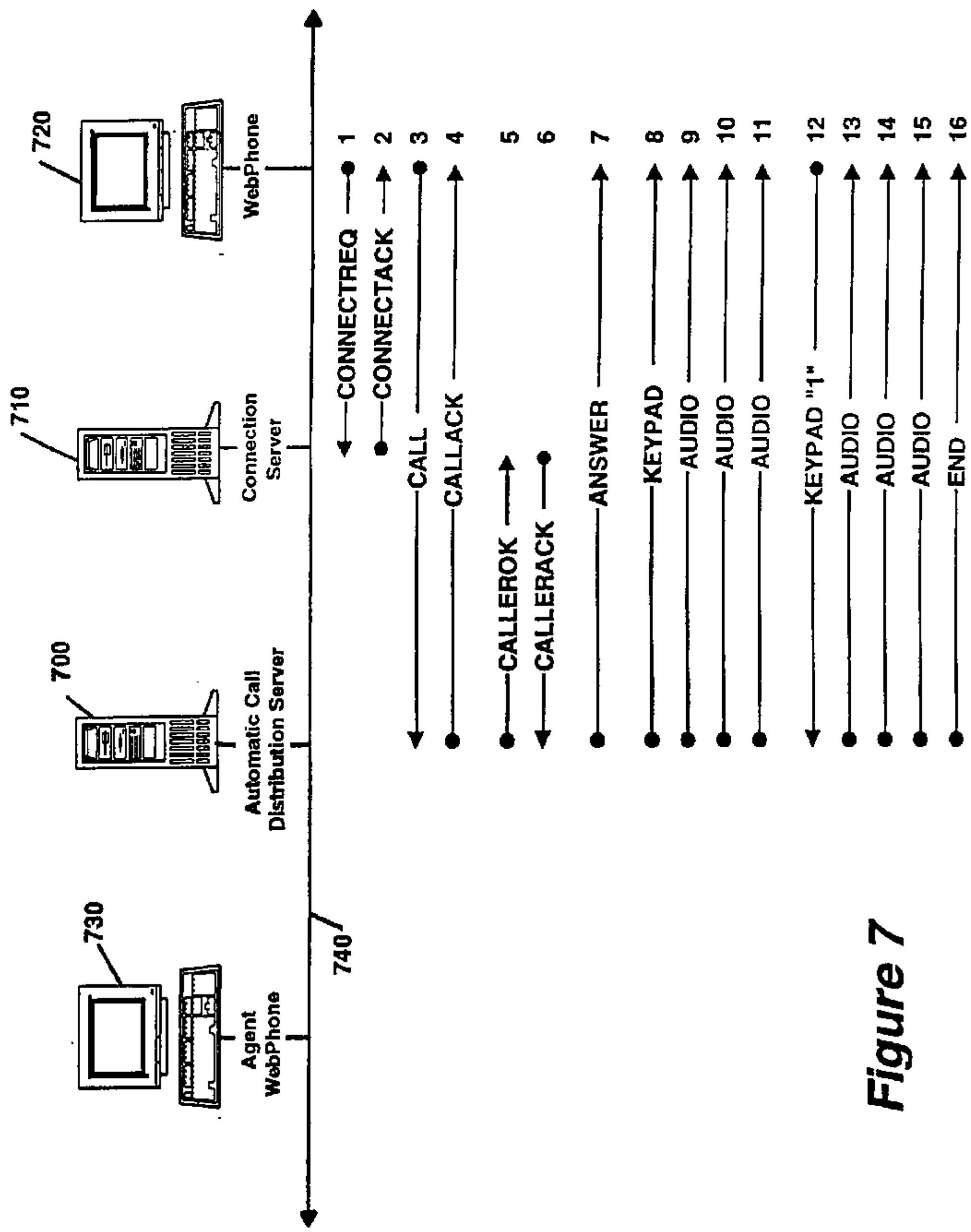
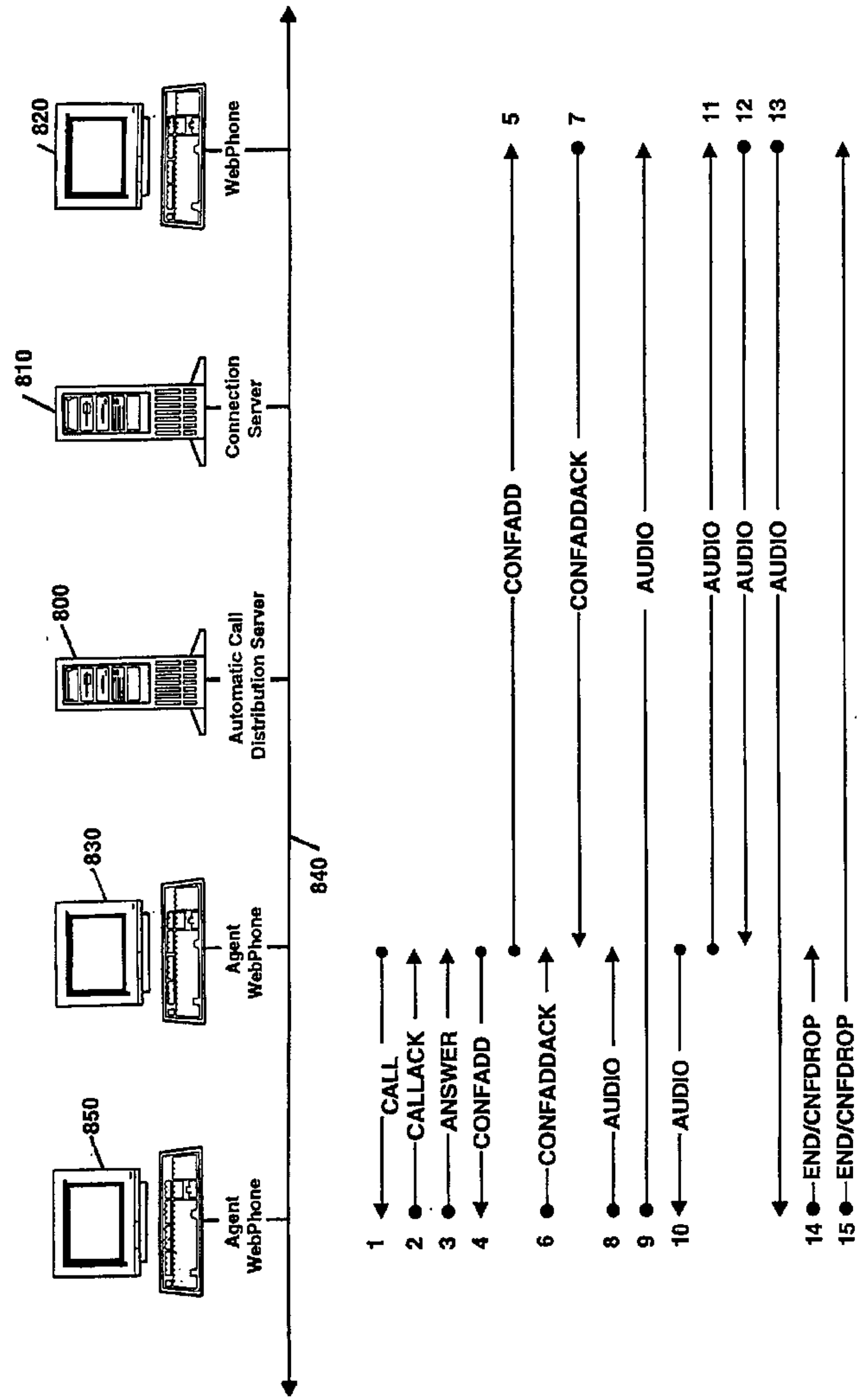

720
WebPhone
710
Connection Server
700
Automatic Call Distribution Server
730
Agent WebPhone
740
CONNECTREQ 1
CONNECTACK 2
CALL 3
CALLACK 4
CALLEROK 5
CALLERACK 6
ANSWER 7
KEYPAD 8
AUDIO 9
AUDIO 10
AUDIO 11
KEYPAD "1" 12
AUDIO 13
AUDIO 14
AUDIO 15
END 16

Figure 7



820
WebPhone
810
Connection Server
800
Automatic Call Distribution Server
830
Agent WebPhone
850
Agent WebPhone
840
1 CALL
2 CALLACK
3 ANSWER
4 CONFADD
5 CONFADD
6 CONFADDACK
7 CONFADDACK
8 AUDIO
9 AUDIO
10 AUDIO
11 AUDIO
12 AUDIO
13 AUDIO
14 END/CNFDROP
15 END/CNFDROP

Figure 8

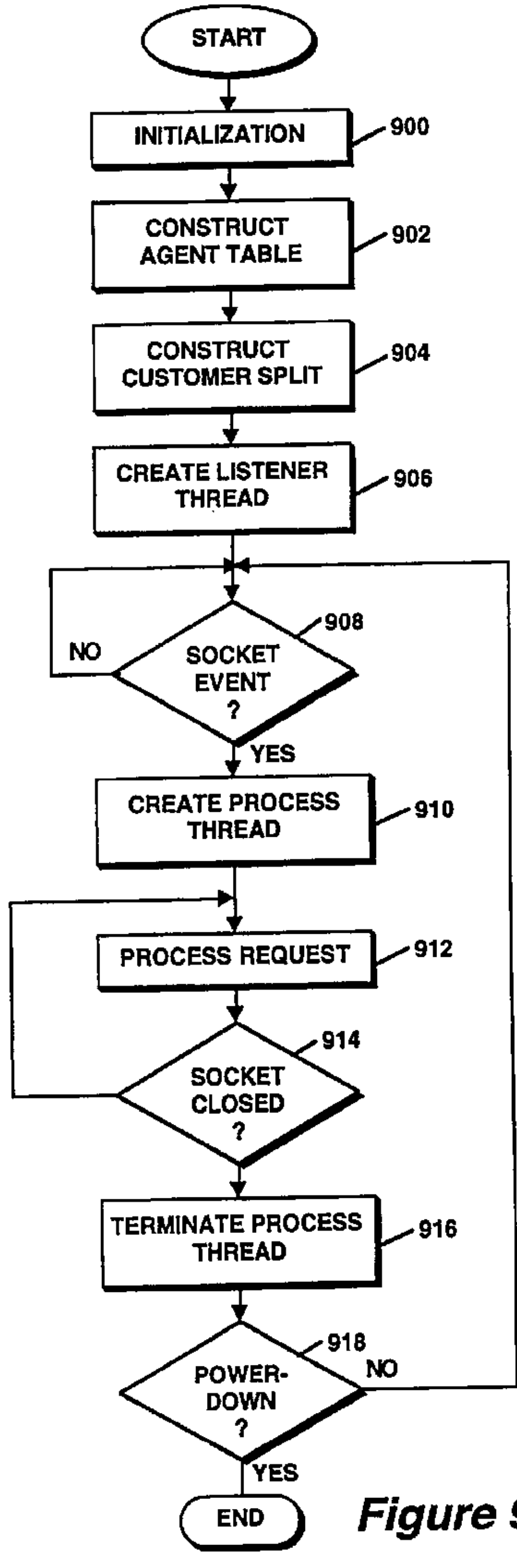

START
INITIALIZATION
900
CONSTRUCT AGENT TABLE
902
CONSTRUCT CUSTOMER SPLIT
904
CREATE LISTENER THREAD
906
NO
SOCKET EVENT ?
908
YES
CREATE PROCESS THREAD
910
PROCESS REQUEST
912
SOCKET CLOSED ?
914
TERMINATE PROCESS THREAD
916
POWER-DOWN ?
918
NO
YES
END

Figure 9

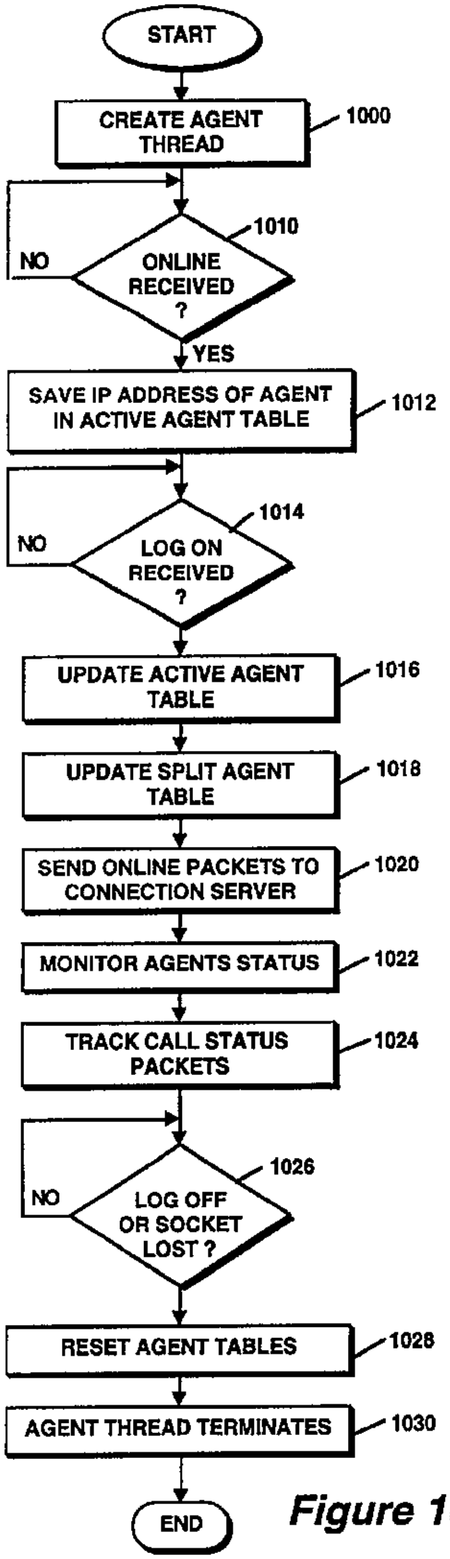
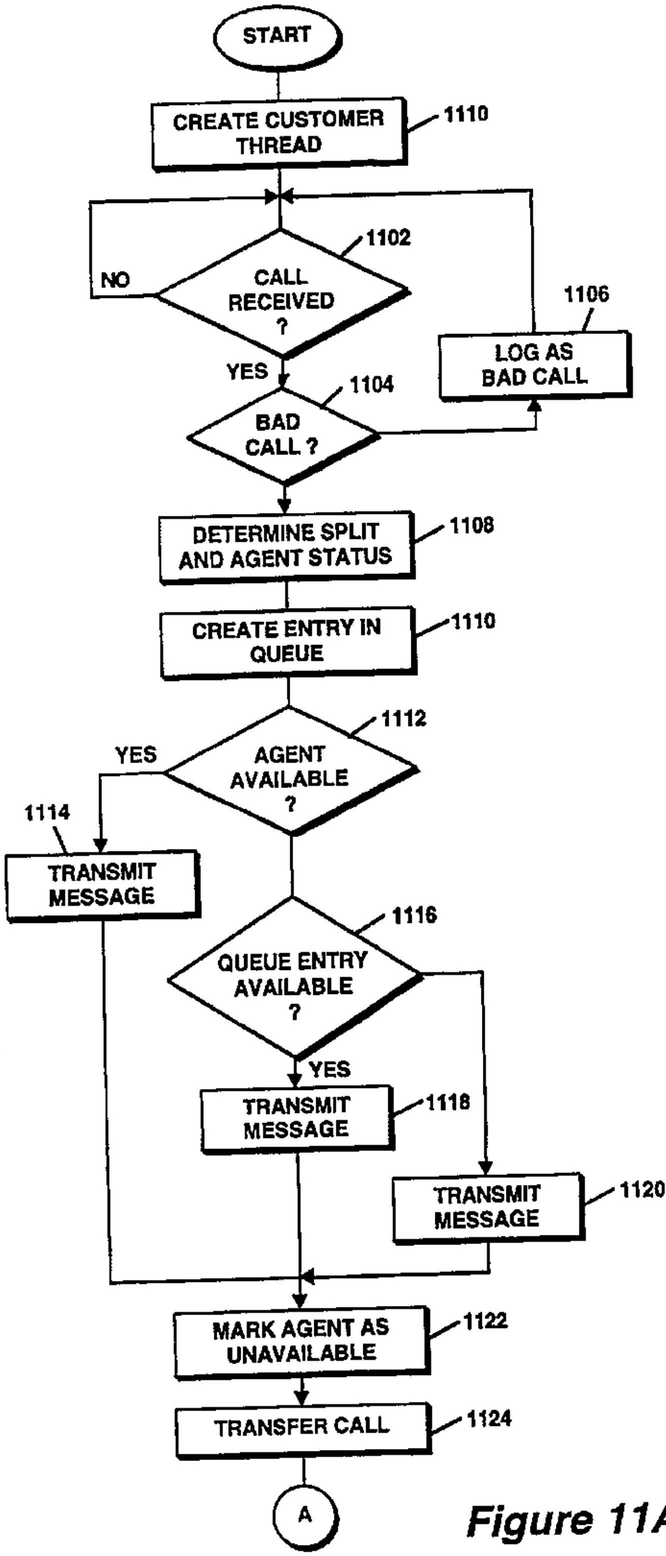

START
CREATE AGENT THREAD
1000
ONLINE RECEIVED ?
1010
NO
YES
SAVE IP ADDRESS OF AGENT IN ACTIVE AGENT TABLE
1012
LOG ON RECEIVED ?
1014
NO
UPDATE ACTIVE AGENT TABLE
1016
UPDATE SPLIT AGENT TABLE
1018
SEND ONLINE PACKETS TO CONNECTION SERVER
1020
MONITOR AGENTS STATUS
1022
TRACK CALL STATUS PACKETS
1024
LOG OFF OR SOCKET LOST ?
1026
NO
RESET AGENT TABLES
1028
AGENT THREAD TERMINATES
1030
END

Figure 10



START
CREATE CUSTOMER THREAD
1110
NO
CALL RECEIVED ?
1102
1106
LOG AS BAD CALL
YES
1104
BAD CALL ?
DETERMINE SPLIT AND AGENT STATUS
1108
CREATE ENTRY IN QUEUE
1110
YES
AGENT AVAILABLE ?
1112
1114
TRANSMIT MESSAGE
QUEUE ENTRY AVAILABLE ?
1116
YES
TRANSMIT MESSAGE
1118
TRANSMIT MESSAGE
1120
MARK AGENT AS UNAVAILABLE
1122
TRANSFER CALL
1124
A

Figure 11A

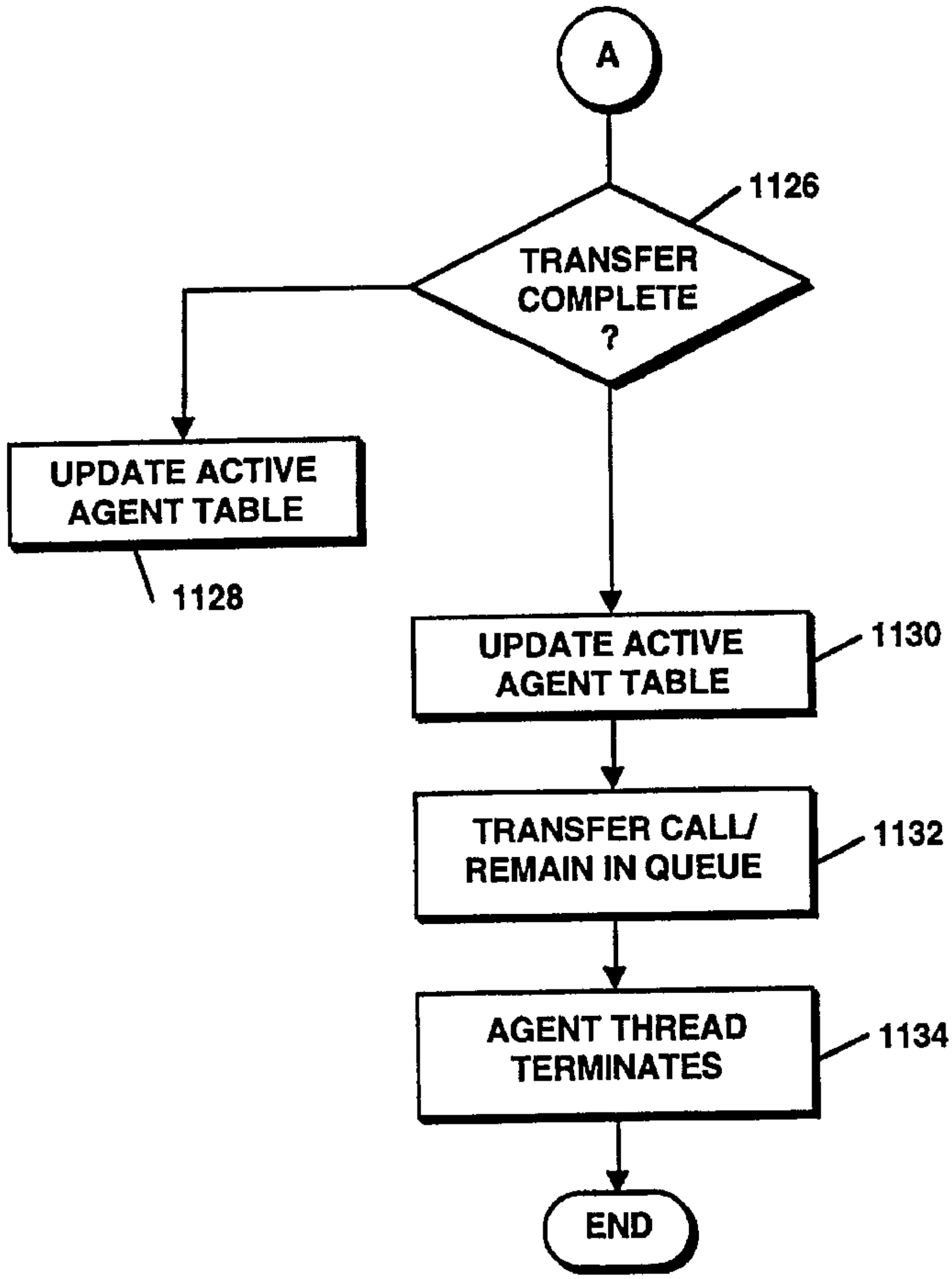

A
TRANSFER COMPLETE ?
1126
UPDATE ACTIVE AGENT TABLE
1128
UPDATE ACTIVE AGENT TABLE
1130
TRANSFER CALL/ REMAIN IN QUEUE
1132
AGENT THREAD TERMINATES
1134
END

Figure 11B

AUTOMATIC CALL DISTRIBUTION SERVER FOR COMPUTER TELEPHONY COMMUNICATIONS

RELATED APPLICATIONS

This application claims priority to United States Provisional Patent Application 60/024,234 entitled WebPhone Automatic Call Distribution Server by Keith C. Kelly, filed Aug. 20, 1996.

In addition, the subject matters of the following related copending applications are incorporated herein by reference:

U.S. patent application Ser. No. 08/533,115 entitled Point-to-Point Internet Protocol, by Glenn W. Hutton, filed Sep. 25, 1995, now abandoned;

U.S. patent application Ser. No. 08/719,894, entitled Directory Server For Providing Dynamically Assigned Network Protocol Addresses, by Mattaway et al., filed Sep. 25, 1996;

U.S. patent application Ser. No. 08/721,316, entitled Graphic User Interface For Internet Telephony Application, by Mattaway et al., filed Sep. 25, 1996;

U.S. patent application Ser. No. 08/719,891, entitled Method And Apparatus For Distribution And Presentation Of Multimedia Data Over A Computer Network, by Mattaway et al., filed Sep. 25, 1996;

U.S. patent application Ser. No. 08/719,554, entitled Point-to-point Computer Network Communication Utility Utilizing Dynamically Assigned Network Protocol Addresses, by Mattaway et al., filed Sep. 25, 1996;

U.S. patent application Ser. No. 08/719,640, entitled Method And Apparatus For Dynamically Defining Data Communication Utilities, by Mattaway et al., filed Sep. 25, 1996;

U.S. patent application Ser. No. 08/719,898, entitled Method And Apparatus For Providing Caller Identification Based Out-going Messages In A Computer Telephony Environment, by Mattaway et al., filed Sep. 25, 1996;

U.S. patent application Ser. No. 08/718,911, entitled Method And Apparatus For Providing Caller Identification Based Call Blocking In A Computer Telephony Environment, by Mattaway et al., filed Sep. 25, 1996;

U.S. patent application Ser. No. 08/719,639, entitled Method And Apparatus For Providing Caller Identification Responses In A Computer Telephony Environment, by Mattaway et al., filed Sep. 25, 1996; and

U.S. patent application Ser. No. 08/832,746, entitled Virtual Circuit Switching Architecture, by Mattaway et al., filed Apr. 4, 1997;

U.S. patent application Ser. No. 08/911,133, entitled Method and Apparatus for Establishing Communications Between Packet-Switched and Circuit-Switched Networks, by Keith C. Kelly, filed Aug. 14, 1997; and

U.S. patent application Ser. No. 08/911,519, entitled Domain Name Server Architecture for Translating Telephone Number Domain Names into Network Protocol Addresses, by Keith C. Kelly, filed August 14, 1997.

FIELD OF THE INVENTION

The invention relates, generally, to data processing systems and telecommunication systems, and, more specifically, to a technique for distributing communications from both circuit-switched networks and packet-switched networks to virtual call centers.

BACKGROUND OF THE INVENTION

Two fundamentally different switching technologies exist that enable digital communications. The first type, circuit-switched networks, operate by establishing a dedicated connection or circuit between two points, similar to public switched telephone networks(PSTN). A telephone call causes a circuit to be established from the originating phone through the local switching office across trunk lines, to a remote switching office and finally to the intended destination telephone. While such circuit is in place, the call is guaranteed a data path for digitized or analog voice signals regardless of other network activity. The second type packet-switched networks, typically connect computers and establish an asynchronous "virtual" channel between two points. In a packet-switched network, data, such as a voice signal, is divided into small pieces called packets which are then multiplexed onto high capacity connections for transmission. Network hardware delivers packets to specific destinations where the packets are reassembled into the original data set. With packet-switched networks, multiple communications among different computers can proceed concurrently with the network connections shared by different pairs of computers concurrently communicating. Packet-switched networks are, however, sensitive to network capacity. If the network becomes overloaded, there is no guarantee that data will be timely delivered. Despite this drawback, packet-switched networks have become quite popular, particularly as part of the Internet and Intranets, due to their cost effectiveness and performance.

In a packet-switched data network one or more common network protocols hide the technological differences between individual portions of the network, making interconnection between portions of the network independent of the underlying hardware and/or software. A popular network protocol, the Transmission Control Protocol/Internet Protocol (TCP/IP) is utilized by the Internet and Intranets>Intranets are private networks such as Local Area Networks (LANs) and Wide Area Networks (WAN). The TCP/IP protocol utilizes universal addressing as well as a software protocol to map the universal addresses into low level machine addresses. For purposes of this discussion, networks which adhere to the TCP/IP protocol will be referred to hereinafter "IP-based" or as utilizing "IP addresses" or "Internet Protocol address".

It is desirable for communications originating from a PSTN network to terminate at equipment in an IP-based network. Problems arise, however, when a user on a circuit-switched network tries to establish a communication link to a packet-switched data network, and vice versa, due to the disparity in addressing techniques among other differences used by the two types of networks. Accordingly, many of the services currently available on network are typically not available to communications originating on the other network.

Automatic call distribution (ACD) centers are one such service which has been used successfully on traditional circuit-switched networks. Typically, a number of human operators or "agents" are used to operate telephones or other terminating apparatus to answer incoming calls for a business entity. Such automatic call centers typically are used by companies which service large numbers of incoming calls for sales, support and customer ordering, etc. Generally, a traditional call center consists of routing and switching hardware and a plurality of terminating equipment located on the same PBX.

With the advent of Internet telephony, the ability to receive incoming communications originating from packet-switched data processing networks, such as the Internet, has given rise for the need to 1) set up similar call distribution facilities for packet-switched calls, 2) adapt existing call

distribution technology to computer telephony and, 3) create automatic call distribution centers which are capable of accepting calls originating from both circuit-switched networks and packet-switched networks.

Further, the success and efficiency of prior automatic call distribution centers to handle the incoming call load is related to the ability to efficiently route incoming communications. As operator reserves change, e.g. shifts change, operators service calls, operator absenteeism, the performance of the system changes dynamically. As such, the physical configuration of the automatic call distribution hardware, and the proximity of the operators and supervisors to operate the hardware, has placed limitations on the flexibility and, therefore, the efficiency with which traditional circuit-switched automatic call distribution systems perform.

Accordingly, a need exists for a system which is capable of implementing traditional automatic call distribution services for communications originating over packet-switched networks.

A further need exists for an automatic call distribution system which is capable of receiving incoming communications from both a packet-switched network and traditional circuit-switched networks.

In addition, a need exists for an automatic call distribution system in which agent operators and/or supervisors may be located in geographically different locations or over different network topologies while still appearing as a single virtual entity.

Yet another need exists for the ability to automatically queue incoming communications based on a number or criteria without requiring user input.

Still a further need exists for the ability to dynamically reorganize the virtual organization of agents within an automatic call distribution system to effectively deal with dynamic call loads and agent resources, etc.

SUMMARY OF THE INVENTION

The Automatic Call Distribution (ACD) server of the present invention provides automatic routing services for calls from both circuit-switched communication networks, such as PSTNs via gateway exchanges, and packet-switched data networks such as the Internet and Intranets. The ACD server allows small businesses and large corporations alike to configure call center agents into virtual groups, departments or support centers and then route incoming communications based on a number of criteria including Caller ID, DNIS, ANI, PBX trunk numbers, first in-first out, longest call on hold, etc. The ACD server of the present invention collects call information and statistics, both incoming and outgoing, and allows call center management to efficiently manage and respond to changes in call load, employee dynamics, employee performance, etc. Call center agents may be physically located in geographically separate locations worldwide and still appear as a single group. Whether operators are working from the home or distant continents, a corporation's sales, marketing, or support department can be reached by the same call and without additional expense. The ACD server of the present invention further provides a graphic user interface that displays 'at a glance' agent load, agent distribution, and call holding queue status enabling center management to drag and drop call center agents between centers, reroute calls to alternate support centers, or enlist the services of agents who can work from home.

According to a first aspect of the present invention, in an automatic call distribution system, a method of distributing

incoming communications over a packet-switched data network, comprises the steps of determining the online status of at least one agent process, defining at least one queue into which incoming communications over the packet-switched network may be placed, each incoming communication containing user information identifying the process from which the communication originated, selectively associating agent processes with the queue in accordance with predetermined criteria, selectively assigning incoming communications to one of the queues in accordance with predetermined criteria, and selectively transferring an incoming communication from the queue to one of the agent processes associated with the queue.

According to a second aspect of the present invention, a computer program product for use with a computer system comprises a computer usable medium having program code embodied in the medium for distributing communications to one or more agent processes, the program code comprising program code means configured to determine at least one agent process operatively coupled to the computer system, program code for defining within the computer system memory a queue, the queue having a plurality of entries, each capable of retaining information associated with an incoming communication, program code, responsive to the agent processes currently online, for enabling association of agent processes with the queue in accordance with a predetermined criteria, program code, responsive to incoming communications, selectively associating an incoming communication with the queue in memory, and program code, responsive to the incoming communications retained in queue and the association of agent processes with the queue, for selectively transferring an incoming communication to an agent process associated with the queue in which the incoming communication information resides.

According to a third aspect of the present invention, an automatic call distribution system for use with a packet-switched data network comprises an automatic call distribution server operatively coupled to the network, a plurality of agent processes operatively coupled to the network, and a control center process operatively coupled to the automatic call distribution server. The automatic call distribution server maintains in a memory thereof a list containing information associated with selected of the agent processes and a list containing information associated with incoming communications. The control center further comprises a graphic user interface for visually displaying and modifying the lists contained within the automatic call distribution server.

BRIEF DESCRIPTION OF THE DRAWINGS

The above and other features, objects, and advantages of the invention will be better understood by referring to the following description in conjunction with the accompanying drawing in which:

FIG. 1 is a block diagram of a computer systems suitable for use with the present invention;

FIG. 2A is a conceptual illustration of a communications network environment in which the present invention may be utilized;

FIG. 2B is a conceptual illustration of a communications network environment in which the present invention may be utilized;

FIG. 3A is a block diagram of an automatic call distribution server system in accordance with the present invention;

FIG. 3B is a conceptual diagram of the data structures for agent tables and incoming call queues in accordance with the present invention;

FIG. 4 is a schematic block diagrams illustrating a packet transfer sequence in accordance with the communication protocol of the present invention;

FIG. 5 is a schematic block diagrams illustrating a packet transfer sequence in accordance with the communication protocol of the present invention;

FIG. 6 is a schematic block diagrams illustrating a packet transfer sequence in accordance with the communication protocol of the present invention;

FIG. 7 is a schematic block diagrams illustrating a packet transfer sequence in accordance with the communication protocol of the present invention;

FIG. 8 a schematic block diagrams illustrating a packet transfer sequence in accordance with the communication protocol of the present invention;

FIG. 9 is a flow chart illustrating the process steps performed by an initialization thread in accordance with the present invention;

FIG. 10 is a flow chart illustrating the process steps performed by an initialization thread in accordance with the present invention; and

FIG. 11A and 11B form a flow chart illustrating the process steps performed by an initialization thread in accordance with the present invention.

DETAILED DESCRIPTION

FIG. 1 illustrates the system architecture for a computer system **100**, such as an IBM PS/2® computer on which the invention can be implemented. The exemplary computer system of FIG. 1 is for descriptive purposes only. Although the description below may refer to terms commonly used in describing particular computer systems, such as an IBM PS/2 computer, the description and concepts equally apply to other systems, including systems having architectures dissimilar to FIG. 1.

The computer system **100** includes a central processing unit (CPU) **105**, which may include a conventional microprocessor, a random access memory (RAM) **110** for temporary storage of information, and a read only memory (ROM) **115** for permanent storage of information. A memory controller **120** is provided for controlling system RAM **110**. A bus controller **125** is provided for controlling bus **130**, and an interrupt controller **135** is used for receiving and processing various interrupt signals from the other system components. Mass storage may be provided by diskette **142**, CD ROM **147** or hard drive **152**. Data and software may be exchanged with computer system **100** via removable media such as diskette **142** and CD ROM **147**. Diskette **142** is insertable into diskette drive **141** which is, in turn, connected to bus **130** by a controller **140**. Similarly, CD ROM **147** is insertable into CD ROM drive **146** which is connected to bus **130** by controller **145**. Hard disk **152** is part of a fixed disk drive **151** which is connected to bus **130** by controller **150**.

User input to computer system **100** may be provided by a number of devices. For example, a keyboard **156** and mouse **157** are connected to bus **130** by controller **155**. An audio transducer **196**, which may act as both a microphone and a speaker, is connected to bus **130** by audio controller **197**, as illustrated. It will be obvious to those reasonably skilled in the art that other input devices such as a pen and/or tablet and a microphone for voice input may be connected to computer system **100** through bus **130** and an appropriate controller/software. DNA controller **160** is provided for performing direct memory access to system RAM **110**. A visual display is generated by video controller **165** which controls video display **170**. Computer system **100** also includes a communications adaptor **190** which allows the system to be interconnected to a local area network (LAN) or a wide area network (WAN), schematically illustrated by bus **191** and network **195**.

Computer system **100** is generally controlled and coordinated by operating system software, such the OS/2® operating system, available from International Business Machines Corporation, Armonk, N.Y. or Windows NT operating system, available from Microsoft Corporation, Redmond, Wash. The operating system controls allocation of system resources and performs tasks such as process scheduling, memory management, and networking and I/O services, among other things. The present invention is intended for use with a multitasking operating system, such as those described above which are capable of simultaneous multiple threads of execution. For purposes of this disclosure a thread can be thought of as a "program" having an instruction or sequence of instructions and a program counter dedicated to the thread. An operating system capable of executing multiple threads simultaneously, therefore, is capable of performing multiple programs simultaneously.

In the illustrative embodiment, an automatic call distribution server in accordance with the present invention is implemented using object-oriented technology and an operating system which supports an execution of an object-oriented programs. For example, the inventive ACD server may be implemented using the C++ language or as well as other object-oriented standards, including the COM specification and OLE 2.0 specification for MicroSoft Corporation, Redmond, Wash., or, the Java programming environment from Sun Microsystems, Redwood, Calif.

Telecommunication Environment

FIG. 2 illustrates a telecommunications environment in which the invention may be practiced such environment being for exemplary purposes only and not to be considered limiting. Network **200** of FIG. 2 illustrates a hybrid telecommunication environment including both a traditional public switched telephone network as well as Internet and Intranet networks and apparatus bridging between the two. The elements illustrated in FIG. 2 are to facilitate an understanding of the invention. Not every element illustrated in FIG. 2 or described herein is necessary for the implementation or the operation of the invention.

A pair of PSTN central offices **210A-B** serve to operatively couple various terminating apparatus through either a circuit switched network or a packet switched network. Specifically, central offices **210A-B** are interconnected by a toll network **260**. Toll network **260** may be implemented as a traditional PSTN network including all of the physical elements including routers, trunk lines, fiber optic cables, etc. Connected to central office **210A** is a traditional telephone terminating apparatus **214** and an Internet telephone **232A**. Terminating apparatus **214** may be implemented with either a digital or analog telephone or any other apparatus capable of receiving a call such as modems, facsimile machines, etc., such apparatus being referred to collectively hereinafter as a terminating apparatus, whether the network actually terminates. Further, the PSTN network may be implemented as either an integrated services digital network (ISDN) or a plain old telephone service (POTS) network. The Internet telephony is conceptually illustrated as a telephone icon symbolizing the Internet telephone client application executing on a personal computer and interconnected

to central office **210A** via a modem **270A**. Similarly, telephone **214C** is connected to central office **210D** and WebPhone **232C** is connected to central office **210B** via modem **270C**. Central offices **210A-B** are, in turn, operatively coupled to Internet **220** by ISP **250B** and **250C**, respectively. In addition, central office **210A** is coupled to ISP**250B** by gateway **218B**. Similarly, central office **210B** is connected to ISP **250C** by gateway **218C**. In addition, a telephone **214B** and Internet telephone **232B**, similar to telephone **214A** and Internet telephone **232A**, respectively, are interconnected to Internet **220** via PBX **212**, gateway **218A** and ISP **258A**. In addition, global server **252** is coupled to the Internet **220** are a domain name system server **254** and **255**. Global server **252** may be implemented as described in U.S. patent application Ser. No. 08/719,894, entitled Directory Server for Providing Dynamically Assigned Network Protocol Addresses, previously referenced and incorporated herein. A global server suitable for use as Global Server **252** is commercially available from NetSpeak Corporation in the form of a collection of intelligent software modules including connection server Part No. CSR1, information server, Model ISR1, and database server, Model DBSR1. Name servers **254** and **255** are described as set forth hereinafter. Finally, Internet Service Providers (ISPs) **250A-D** may comprise any number of currently commercially available Internet service providers such as America On Line, the IBM Global Network, etc. An Intranet implemented as LAN **275** is coupled to Internet **220** via ISP **250D** and server **256**. Server **256** may have the architecture as illustrated in FIG. **1** and functions as a proxy server for LAN **275** to which WebPhone **232E** is connected via a LAN-based TCP/IP network connector **280**. A plurality of Internet telephone **232F** and **232E** are coupled to LAN **275** via LAN connectors **280**. The gateways and Internet telephony client applications may be implemented as set forth in greater detail hereinafter.

WebPhone Client

Internet telephone **232** may be implemented as described in the previously referenced U.S. patent applications incorporated herein by reference. An Internet telephony application suitable for use with the present invention is the WebPhone 1.0, 2.0 or 3.0, client software application commercially available from NetSpeak Corporation, Boca Raton, Fla., referred to hereafter as the WebPhone client. For the remainder of this description, the Internet telephone will be referred to as the WebPhone client. It will be obvious to those reasonably skilled in the arts that other Internet telephone applications implementing similar functionality may be substituted for the WebPhone without affecting the inventive concepts contained herein. The WebPhone client comprises a collection of intelligent software modules which perform a broad range of Internet telephony functions. For the purpose of this disclosure, a "virtual" WebPhone client refers to the same functionality embodied in the WebPhone client application without a graphic user interface. Such virtual WebPhone client can be embedded into a gateway, automatic call distribution, server, or other apparatus which do not require extensive visual input/output from a user and may interact with any other WebPhone clients or servers adhering to the WebPhone protocol. For the purpose of this disclosure, WebPhone client **232** or any of the virtual WebPhone clients may be implemented in other apparatus, may be considered WebPhone client applications, "WebPhone Clients", as opposed to other apparatus such as the connection/information server, which adheres to the WebPhone Protocol.

The WebPhone software applications may run on the computer system described with reference to FIG. 1, or a similar architecture whether implemented as a personal computer or dedicated server. In such an environment, the sound card **197** accompanying the computer system **100** of FIG. 1, may be an MCI compliant sound card while communication controller **190** may be implemented through either an analog modem **270** or a LAN-based TCP/IP network connector **280** to enable Internet/Intranet connectivity.

The WebPhone clients, as well as any other apparatus having a virtual WebPhone embodied therein, each have their own unique E-mail address and adhere to the WebPhone Protocol and packet definitions, as extensively described in the previously referenced related U.S. patent applications. For the reader's benefit, short summary of a portion of the WebPhone Protocol is set forth to illustrate the interaction of WebPhone clients with each other and the connection/information server when establishing a communication connection.

Each WebPhone client, may serve either as a calling party or a caller party, i.e. the party being called. The calling party transmits an on-line request packet to a connection/information server upon connection to an IP-based network, e.g. the Internet or an Intranet. The on-line request packet contains configuration and settings information, a unique E-mail address and a fixed or dynamically assigned IP address for the WebPhone client. The callee party, also a utilizing a WebPhone client, transmits a similar on-line request packet containing its respective configuration and setting information, E-mail address and IP address to the same or a different connection server upon connection to an IP-based network. The calling party originates a call by locating the callee party in a directory associated with either its own WebPhone client or the connection/information server to which it is connected. The callee party may be identified by alias, E-mail address or key word search criteria. Once the E-mail address of the calling party is identified, the calling party's WebPhone forwards a request packet to the connection/information server, the request packet containing the callee party's E-mail address. The connection/information server uses the E-mail address in the received request packet to locate the last known IP address assigned to the callee party. The connection/information server then transmits to the calling party an information packet containing the IP address of the callee party. Upon receipt of the located IP address from the connection server, the calling party's WebPhone client initiates a direct point-to-point communication link with the callee party by sending a call packet directly to the IP address of the callee party. The callee party either accepts or rejects the call with appropriate response packets. If the call is accepted, a communication session is established directly between the caller and the callee, without intervention of the connection/information server. The above scenario describes establishment of a communication link which originates and terminates with clients on an IP-based network.

To facilitate interaction with WebPhone clients, a virtual WebPhone is implemented in the gateway **218**, either executable in RAM memory or embedded in ROM memory associated with such apparatus. The gateway **218** comprises a virtual WebPhone client which acts as a proxy device and voice processing hardware that bridges from an IP-based network to a PSTN network. The gateway **218** may be implemented with either a microprocessor based architecture or with dedicated digital signal processing logic and embedded software. A gateway suitable for use as gateway **218** with the present invention is either NetSpeak Model Nos. WGX-MD/24, a 24-port digital T-1 IP telephony

gateway, or WGX-M/16, a 16-port analog IP telephony gateway, both commercially available from NetSpeak Corporation, Boca Raton, Fla. Gateway **218** is described in greater detail with reference to FIG. **3** hereinafter.

One of the capabilities of the gateway **218** is to bridge between the PSTN and Internet/Intranet, and the Internet/ Intranet and the PSTN. Gateway **218** virtualizes the PSTN call, making it appear as just another WebPhone client call. This virtual WebPhone process interfaces with ACD server **242** so that incoming PSTN calls can be routed to agent WebPhone processes with the tracking, distribution, and monitoring features of the ACD server **2424**, as described hereinafter. For incoming calls originating on a PSTN, gateway **218** provides to ACD server **242** information about incoming calls so that proper call routing can ensue, such information possibly comprising Caller ID (CLID), automatic number identification (ANI), DNIS, PBX trunk information, from the central office **210**, or other information collected by voice response units.

A communication link over a packet-switched network may be established with the network illustrated in FIG. **2A**, using the WebPhone protocol as disclosed in U.S. patent application Ser. No. 08/533,115 entitled "POINT-TO-POINT INTERNET PROTOCOL" by Glenn W. Hutton, filed Sep. 25, 1995, previously incorporated herein by reference. Specifically, WebPhone **232A** may connect to Internet **220** through central office **210A**, ISP **250B** and register with global server **252**, notifying server **252** of its current dynamically signed Internet protocol address. Subsequently, WebPhone client **232A** may inquire as to the current Internet protocol address of another WebPhone client, for example, WebPhone client **232C**. If WebPhone client **252** is currently connected to the Internet and has likewise registered with the global server **252** will return the Internet protocol address of WebPhone **232C** to WebPhone **232A**. WebPhone client **232A** may then establish a direct connection to WebPhone client **232C** via central office **210A**, ISP **250B**, Internet **220**, ISP **250C**, and central office **210B**. Alternatively, a point-to-point connection over a packet-switched network may be established over a local area network **275** by means of a direct connection from WebPhone clients **232E** to **232F**, such connection being possible if the Internet protocol addresses of the respective WebPhones are fixed.

Having explained the telecommunication environment and a number of possible interactions between terminating apparatus on a circuit-switched network and executing tasks on a packet-switched network, a simplified telecommunication environment is illustrated in FIG. **2** to facilitate an understanding of the invention. Where possible, the same reference numbers are utilized in FIG. **2B** as in FIG. **2A**.

Referring to FIG. **2B**, a pair of terminating apparatus **214A-B** are coupled to a public switched telephone network central carrier office **210** which is, in turn, coupled to a public branch exchange/public access branch exchange (PBX/PABX) **212**, in a manner as previously described with reference to FIG. **2A**. A second pair of terminating apparatus, i.e. telephones **214C-D**, are connected directly to PBX/PABX **212**, as illustrated. PBX/PABA**212** is coupled to a gateway **218** which is, in turn, coupled to a packet-switched data network, illustrated as Internet/Intranet **220**. Coupled to network **220** are automatic call distribution server **242**, a connection server **252**, one or more caller WebPhone client processes **232**, and one or more agent WebPhone client processes **244A-F**. Gateway **218**, connection server **252**, Internet **220**, and WebPhone clients **232** and **244** have the structure and functionality as previously described herein or as referenced in copending application. In addition, a control center **280**, host system **260**, and conference server **270** are operatively coupled to ACD server **242** over network **220**.

Host system **260** may be used in conjunction with the ACD system of the present invention but is not an integral part thereof. Host system **260** typically comprises a server and database on which customer records, such as name address, product registration information, warranty information, etc. are stored. Host system **260** may be coupled to ACD server **242** over a private or a public network. In order for agent processes working from satellite locations to receive, review, and update customer information, ACD server **242** may notify optional host system **260** of a received customer call based on available information, i.e. ANI, CLID, DNIS, etc. Host system **260** then generates an HTML document that may be retrieved by the agent process when the call is routed to the agent. The agent may then review, modify, and upload the HTML data to the host. Alternatively, the host generates an HTML password protected page on a server such as a Web Server. The agent process is sent the URL and password to the Web page. The agent may then access the protected page with a secure browser. The agent may then review, modify, and upload the HTML data.

Conference server **270** may be used in conjuntion with the ACD system of the present invention but is not an integral part thereof. Conference server **270** typically comprises a server, network interface logic and logic necessary to replicate data, such as AUDIO and/or VIDEO packets, for transmission over a packet-switched network in accordance with the WebPhone protocol. Conference server **270** may be coupled to ACD server **242** over a private or a public network. In the ACD server **300** environment, it is advantageous for agents to call supervisors when situations with customers warrant assistance. After conferencing in a supervisor, agents need the capability to stay on the line until the supervisor drops off, or allow the supervisor to exit from the call and finish call processing themselves. Additionally, supervisors need the ability to call into conversations in progress and monitor agent performance and quality of service. In certain instances, however, a conference server is not required, particularly were the number of parties is small. In such instances the conferencing functionality embodied in the WebPhone protocal, as described with reference to FIG. **8** is sufficent.

ACD server **242**, control center **280**, and agent WebPhone clients collectively function as the ACD system of the present invention. The structure and function of ACD server **242** and control center **280** are described hereinafter with reference to FIGS. **3-11**.

Automatic Call Distribution Server Architecture

FIG. **3A** illustrates conceptually the system architecture of ACD Server **300**, which may be used as ACD server **242** of FIG. **2B**. ACD Server **300** may be implemented using computer architecture similar to computer system **100** as described in FIG. **1** such elements having been described in detail and not shown in FIG. **3**. ACD server **300** comprises multiple software modules which collectively enable all facets of call progress and call handling to be controlled by ACD server **300**. Specifically, ACD Server **300** comprises a graphic user interface **302**, control center module **304**, voice response module **306**, call routing module **308**, network interface **310**, WebPhone client **314**, a memory **316** and an optional database **312**.

As illustrated in FIG. 2B, although agent processes **244** comprise portions of the automatic call distribution system, they may be located remotely from ACD server **300** over either a private or global packet-switched network. Similarly, control center **304** and GUI **302**, although illustrated in FIG. 3B as part of ACD server **300** may likewise be located remotely from ACD server **300** over either a private or global packet-switched network. The control center **304**, agent WebPhone processes **244** and ACD server **300** with one another using the various packets of the WebPhone protocol, as described herein and in the previously referenced copending applications. In this manner, a virtual automatic call distribution system may comprise a supervisor at a control center and one or more agents all of whom may be located in geographically separate locations, but are connected over a packet-switched data network to the ACD server **300**, and collectively appear to an incoming caller as a single virtual entity.

The ACD Server **300** resides logically between the agent WebPhone processes **244** that form the business unit utilizing automatic call distribution, and a global or local connection server **252**. ACD server **300** appears as a series of virtual WebPhone processes to connection server **252**, one WebPhone process for each group ACD server **300** represents, each with unlimited line capability. Agent processes **244** are configured with the ACD server **300** as their respective connection server, however, ACD server **300** does not actually duplicate the connection server functionality. Instead, ACD server **300** may be configured to invoke the services of connection server **252**, as needed. Necessary packets and required information from agent WebPhone **244** are reflected or routed to connection server **252** by ACD server **300** thereby enabling ACD server **300** to perform call tracking and statistics collection, as required.

Control Center Module

Although illustrated in FIG. 3A as part of a CD server **300**, control center **304** and graphic user interface **302** may be implemented as a remote module couplable to ACD server **300** over a network. Control center **304** provides a graphic user interface **302** and a plurality of options and dialog boxes through which the control center may send packetized commands to ACD server **300**. Specifically, graphic user interface **302** may be implemented using standard Windows APIs to display the previously described tables and queues in tabular or iconic form. In the manner reasonably understood by those skilled in the programming arts, information within the tables and queues may be dragged and dropped using a mouse or other pointer device with graphic user interface **302** to manipulate the contents of the tables. Changes to the tables and/or parameters of the queues are then sent in packetized form to ACD server **300** which appropriately updates the memory **316** or optional database **312**. Control center **304** enables the supervisor user to set queue depth, reassign agents to groups or splits, track call progress, etc. A number of packets utilized by control center **304** to communicate commands and data to ACD server **300** are described hereinafter.

In the illustrative embodiment, GUI **302** in control center **304** is designed to allow viewing of a single department at a time. When the Control Center **304** is launched, queries ACD server **300** to supply current status of the department being viewed and subsequently send only "events" that occur within a department. Control center **304** will track the time between these events and update the GUI **302** accordingly. Examples of "events" packets iinitiaited by ACD server **300** upon the occurence of:

New call arrives in queue.

Agent finishes call and hangs up.

Agent Enters Do Not Disturb Mode.

Agent Leaves Do Not Disturb Mode.

Agent Goes Off-line.

Agent Goes Online.

Agent Picks Up Personal Line.

Agent Hangs Up Personal Line.

The control center user may desire to see the agent WebPhone processes per department or only those WebPhone processes ONLINE. Each of the control center initiated packet specifies the options desired.

ACD server **300** further comprises a call response module **306** which functions to provide periodic responses to incoming calls. The construction and function of call response module **306** is similar to that of a regular WebPhone client without a graphic user interface. In addition, one or more prerecorded messages in the form of packetized audio and/or video and text data may be stored in a local memory or within memory **16**, or possibly database **312**. The call response module transmits messages to incoming calls as determined by the appropriate customer or agent threads, as described herein.

Call Router Module

ACD server **300** further comprises a call router module **308** which receives parameter data from control center **304** and oversees the creation of appropriate threads to route calls effectively. In the illustrative embodiment, a number of different criteria may be utilized to route incoming calls, several such algorithms are described as follows:

Call Control Vectoring (CCV) on Split Basis—Call Control Vectoring allows the user to configure their ACD to route calls to various queues or splits. Each split may be configured with its own call routing algorithms.

CCV based on Queue Depth—Support for CCV based on queue statistics. Queues reaching certain levels of activity or call hold time to Agent on duty ratios result in alternate CCV to other backup call centers, announcements, or backup/alternate agent strategies.

Stranded Call Routing—Calls left on the queue for excessive lengths of time can be automatically routed to receive announcements or special Agents.

CCV based on Assigned Agent Priority—Agents assigned to a particular split can be additionally assigned a priority that influences CCV. A collection of Agents assigned the same priority are routed calls equally. As split depths increase, lower priority agents from other splits, working at remote sites, working from home, etc. can be routed calls as needed to satisfy customer needs.

Language Support/Call Routing—CCV support for call routing based on language requirements.

Time of Day and Week CCV—Alternate CCV based on the time of day or day of week to support automatic load balancing. A call center receiving calls after 5:00 pm EST could be route automatically to PST call centers.

The actual routing algorithms may be implemented a modular routines in memory **316** indexed by indentifiers and specifiable by specific commands from the call center identifying the routine by which calls are to be answered.

Call Router Module **308** of ACD server **300** reroutes packets in the same manner as connection server **252** enabling ACD server **300** to act as a virtual connection server without having to duplicate all of the connection

server functionality. ACD server **300** acts as a virtual WebPhone client for each of the incoming call. Incoming calls may originate from other addresses on a packet-switched network or from terminating apparatus on a circuit-switched network. As an incoming call is received, ACD server **300** may accepts the call conditionally on queue depth and configuration information received from control center **304**. For each call, an entry representing the call is placed into one of the queue(s) and two network sockets are opened. One socket functions to transmit audio and/or other data while the other socket functions to transmit control data. The number of sockets, therefore, partially determines the number of incoming calls that may be opened at any instant.

Table and Queue Structures

ACD server **300**, upon initialization, sets up a number of tables and queues necessary for routing and tracking of incoming calls, as illustrated by FIG. 3B. The automatic call distribution system of the present invention is intended to service calls for one or more groups, such as sales and support within a single organization or for multiple organizations. Such groups are referred to hereinafter as splits or queues. The number of splits as well as the agents assigned to a split and the call routing algorithm to be used with calls within a split queue are defined using the control center module **304**, as explained herein.

Referring to FIG. 3B, a department list **320** is maintained within memory **316** by ACD server **300**. The department list **320** includes information identifying each of the groups or splits for which the ACD currently has responsibility, as defined by the ACD system user, hereinafter referred to as a supervisor. Department list **320** may be implemented as a linked list or a doubly-linked list or, alternatively, may be implemented as a series of records within database **312**. The information retained regarding each department may comprise a department identifier, the E-mail address of the department and an Internet protocol address, if a fixed address has been assigned to the department.

ACD server **300** also maintains in memory **316** or in optional database **312** an Agent Information Table **330**, as illustrated in FIG. 3B. Each entry of the Agent Information Table includes an agent identifier by which the system may track agents, as well as user information for each agent operator associated with the automatic control distribution system. In the contemplated invention, an agent is a human operator operating WebPhone client software on a computer architecture, such as that illustrated in FIG. **1** and which is operatively coupled to a packet-switched data network to ACD server **300**. Each entry of the Agent Information Table **330** may further include supplemental identification data such as that contained in the user information packet of the WebPhone protocol, e.g. agent name, address, E-mail address, etc. Where large numbers of agents are employed in connection with an automatic call distribution system, the Agent Information Table **330** may be implemented as a series of records in database **312** with each agent having one or more records associated therewith. A separate Agent Information Table **330** may be compiled for each entry of department list **320**.

An Active Agent Table **340** is utilized by ACD server **300** to continuously monitor the status of agents. Active Agent Table **340** may be implemented as a circular list in memory **316**. Each entry of table **340** includes an agent identifier, a status indicator, and the current Internet protocol address assigned to the agent's WebPhone process. In the illustrative embodiment, to prevent simultaneous modification of an

entry in a multithreaded environment, either a single entry or the entire table **340** may be accessed only through a mutually exclusive semaphore mechanism, such mechanisms known within the programming arts. In this manner, an agent's status is maintained current. As with tables **320** and **340**, the Active Agent Table **340** may be implemented using linked lists within memory **316** or, alternatively, may be implemented using optional database **312**.

ACD server **300** further constructs a Split Agent Table **350** whose entries contain information on agents who are actively online and to one or more spilts or groups from which they may receive incoming calls. As with tables **320** and **330**, table **350** may be implemented as a linked list or doubly linked list, or, alternatively, may be implemented in optional database **312**. The interaction of tables **330**, **340** and **350** is described with reference to FIGS. **9–11**.

In addition, ACD server **300** constructs a queue **360** for each group for which ACD server **300** has responsibility. Once an incoming call is received by ACD server **300**, a call record is created and a thread spawned to manage the record, as described hereinafter. The entries of queue **360** include the call record for the queue. The call record is updated as the call progresses, e.g. messages are played, connection lost, etc. Once a record has progressed to the top of a queue, the call will be assigned to an agent. In addition to maintaining the status of a call, the call record may also include information, such as user information contained within a USERINFO packet identifying the caller. Queue **360** is implemented in memory **316** in the illustrative embodiment as a priority queue data structure. In the illustrative embodiment, such priority queue is not a pure First In First Out (FIFO) queue, but allows each entry to be assigned a priority which causes some entries to be handled prior to their actual position and queue. If all entries in a priority queue were provided with the same priority number, a pure FIFO queue would result. In this manner, certain entries may be taken for others, however, the general protocol is that the oldest calls are serviced first.

ACD Packet Transfer Sequences

In FIGS. **4–8**, agent and caller process may be implemented as WebPhone client processes. The network may be any combination of LAN and WAN network technologies, i.e. the Internet or Intranet. Further, the connection server may be connected to the ACD server locally over a private network, i.e. an Intranet or globally over a global network, i.e. the Intranet. The agent process may likewise be connected to the ACD server either globally or locally. Additionally, the caller processes may originate on packet-switched networks as well as circuit-switched networks such as PSTNs using a gateway exchange (not shown). The format and description of the packets referred to with reference to FIGS. **4–8** are described in detail herein as well as with reference to the WebPhone Protocol in the previously referenced copending, related patent applications.

FIG. **4** illustrates schematically the packet transfer sequence between a caller process **430**, a connection server **410**, an ACD server **400** and an agent process **430**, over networks **440**. Specifically, the packet sequence illustrated in FIG. **4** illustrates the initialization process performed by ACD server **400** upon power on and testing. Following power on and testing of its architecture, ACD server **400** connects to a packet-switched data network **440** and transmits to connection server **410** an ONLINE packet for one of the departments, i.e. split groups which ACD server **400** is configured to represent, as illustrated by transmission **1**. For

example, transmission 1 may be the ONLINE packet for the sales department of a particular company having an E-mail address "sales @ company.com." In response, connection server 410 transmits and ONLINEACK packet to ACD server 400, as illustrated by transmission 2. ACD server 400 continues the above process for each department for which ACD server 400 is currently configured to represent. For example, transmissions 3 and 4 of FIG. 4 may represent the ONLINE packet and ONLINE acknowledge packets, respectively, for the support organization, having an E-mail address "support @ company.com."

Next, ACD server 400 processes Agent Information Table 330 from database 312 and creates entries at the connection server 410 for each WebPhone agent process having telephone privileges. This process is represented by transmission of a USERINFO packet, illustrated as transmission 5. The USERINFO packet contains the E-mail address of the respective agent, the agent name, etc. In response, connection server 410 transmits to ACD server 400 a USERINFO acknowledge packet, illustrated as transmission 6. Next, ACD server 400 transmits an ONLINE packet to connection erver 410, as illustrated by transmission 7 for the specific agent. In response, connection server 410 transmits an ONLINEACK packet to ACD server 400, as illustrated by transmission 8.

Once ACD 400 is online and has transmitted, an appropriate USERINFO packet for each agent to connection server 410, the respective agents 430 may log on to ACD server 400 as follows. An agent transmits an ONLINE packet to ACD server 400, as illustrated by transmission 9. In response, ACD server 400 transmits an ONLINEACK packet to agent 430, as illustrated by transmission 10. This process occurs for subsequent agents, as illustrated by transmissions 11 and 12. Agent 430 then transmits a LOGON packet to ACD server 400, as illustrated by transmission 13. In response, ACD server 400 transmits to agent 430 a LOGONACK packet, illustrated as transmission 14. The LOGON packet for an agent 430 may include the agent number and a password. Once the LOGONACK packet is received from ACD server 400, the agent is able to accept incoming calls. This process is continued for all agents 430 currently online, as illustrated by transmissions 15 and 16. For agents 530 already online at the time of initialization of ACD server 400, a PULSE packet (not shown) transmitted from agent 430 to ACD server 400 will achieve the same result as transmissions 9–16 of FIG. 4. A PULSE packet is described in the previously referenced copending applications.

Following the packet transmission sequence described with reference to FIG. 4, ACD server 400 is initialized with its respective connection server 410 and the respective agents 430 associated with ACD server 400 are similarly initialized and ready to receive calls.

FIG. 5 illustrates schematically a packet transfer sequence between a caller process 520, a global server 510, an ACD server 500 and an agent process 530 over network 550. For the scenario illustrated by FIG. 5, it may be presumed that ACD server 500 has no voice mail functionality or any voice mail functionality, is disabled, and further, the split queue within the ACD server 300 is not full. First, caller process 520 transmits to connection server 510 a CONNECTREQ package, as illustrated by transmission 1. The format and content of all packets illustrated in FIG. 5 are described in detail in the previously referenced patent applications. In response, connection server 510 transmits a CONNECTACK packet to caller process 520, as illustrated by transmission 2. With this exchange of packets, the caller process 520 is able to obtain a reference to the appropriate group organization serviced by ACD server 500, such as "sales @ company.com" and is provided with the Internet Protocol address of the ACD server 500 by connection server 510. Next, caller process 520 transmits a CALL packet to ACD server 500, as illustrated by transmission 3, thereby initiating a call to the ACD server 500. In response, ACD server 500 transmits a CALLACK packet to caller process 520, as illustrated by transmission 4. Next, ACD server 500 verifies the validity of the caller process 520 by transmitting a CALLLEROK packet to connection server 510, as illustrated by transmission 5. In response, connection server 510 transmits a CALLERACK packet to ACD server 500, as illustrated by transmission 6, indicating that the caller process 520 is valid. The caller validation process and the utilization of the CALLEROK and CALLERACK packet is described in detail in U.S. patent application Ser. No. 08/719,894 entitled DIRECTORY SERVER FOR PROVIDING DYNAMICALLY ASSIGNED NETWORK PROTOCOL ADDRESSES with regard to FIG. 17A thereof.

Next, if the split queues within ACD server 500 are not full, ACD server 500 sends an ANSWER packet to caller process 500, as illustrated by transmission 7, and places the call on the "sales" queue for servicing by an agent 530.

While a caller process 520 is in queue, ACD server 500 enables the caller process keypad and may transmit periodic audio messages such as "Please wait for next available agent or press 1 to request a callback," and/or other information such as music, advertisements, stock quotes, etc. This process is illustrated in FIG. 5 by transmission of a KEYPAD packet from ACD server 500 to caller process 520, as illustrated by transmission 8. The subsequent audio message and/or audio sound track is illustrated by transmission of one or more AUDIO packets from ACD server 500 to caller process 520, as illustrated by transmissions 9–11. Although such messages are illustrated as transmissions of audio data, the information may conveyed as video, text, or multimedia data using the appropriate packet types and the WebPhone client software. If the caller presses "1" to request a callback, a KEYPAD "1" packet is transmitted from caller 520 to ACD server 500, as illustrated by transmission 12. In response, ACD server 500 may transmit one or more AUDIO packets to caller process 520 representing a message such as "Thank you for calling, a representative will return your call," as illustrated by transmissions 13–15 of FIG. 5. Finally, an END packet is transmitted from ACD server 500 to caller process 520, as illustrated by transmission 16, terminating the communication.

Alternatively, if an agent is available and the caller has not requested a callback, the call is transferred to an agent with the transmission of a TRANSFERREQ packet from ACD server 500 to caller 520, as illustrated by transmission packet 17. A CALL packet is then transmitted from caller 520 to agent 530, as illustrated by FIG. 18. Agent 530 then transmits a CALLACK packet to caller 520 as illustrated by transmission 19. Alternatively, if agent 530 is busy, any of the ANSWER, ANSWER MACHINE, BUSY, REJECT, or ERROR packets may be transmitted back to caller 520, as illustrated by transmission 20. In response, caller 520 transmits a TRANSFERACK packet to ACD server 500, as illustrated by transmission 21. ACD server 500 transmits a TRANSFER packet to caller 520, as illustrated by transmission 22, to facilitate transfer of the call to another agent process 530. Alternatively, if agent 530 was available to accept the call, transmission 19 would be followed by multiple AUDIO packets between agent 530 and caller 520, such packets containing the audio content of the communi-

cation between the parties. Such exchange of AUDIO packets as well as other types of packets including VIDEO, TEXT, etc. continues until either party transmits an END packet, similar to transmission **16**.

FIG. **6** illustrates the interaction between a caller **620**, a global server **610**, an ACD server **600**, and an agent **630** interconnected over network **640**, similar to the scenario described with reference to FIG. **5**, with the exception that ACD server **600** has voice mail functionality which is enabled. Accordingly, transmissions **1–12** of FIG. **6** are identical to transmissions **1–12** of FIG. **5**. In this scenario, since voice mail is enabled, the AUDIO packet transmitted from ACD server **600** to caller **620**, as illustrated by transmission **13**, may comprise a message such as "Please leave your message at the tone" or other audio queue. In response, caller **620** may transmit one or more AUDIO packets and an END packet, as illustrated by transmissions **14–16**, of FIG. **6**. Alternatively, ACD server **600** may transmit additional AUDIO packets representing a message such as "Thank you for calling, a representative will return your call," followed by a END packet, such transmissions not illustrated in FIG. **6**. The remaining packet sequence illustrated by transmissions **17–22** of FIG. **6** is similar to that of FIG. **5** as well.

The packet transfer sequences illustrated in FIGS. **5** and **6** occur when agents are available to respond to calls from caller processes. If no agents are available, the packet transfer sequence between caller **720**, connection server **710**, ACD server **700** and agents **730** occurs, as illustrated in FIG. **7**. In FIG. **7**, transmissions **1–7** are identical to transmissions **1–7** of FIGS. **5** and **6**, except that ACD server **700** does not place an incoming call into its respective queue. If ACD server **500** has voicemail disabled, a packet transfer sequence, illustrated by transmissions **8–16** of FIG. **7** will occur, similar to transmissions **8–16** of FIG. **5**. Thereafter, however, since an agent is not available, transmission **17–22** which occurred with reference to FIG. **5**, will not occur with reference to the scenario illustrated by FIG. **7**.

FIG. **8** illustrates schematically the packet transfer sequence between a caller process **830**, an ACD server **800**, an agent process **830**, and a second agent process **850** over network **840**. For the scenario illustrated by FIG. **8**, it is assumed that a call connection has been established between caller process **820** and agent process **830**, the packet transfer sequence related to establishment of such a call not shown in light of FIGS. **4–7**. The agent now wishes to conference a second agent, e.g. the supervisor, utilizing the conferencing functionality contained within the WebPhone client software versions 2.0 and thereafter. The first, agent process **830**, transmits a CALL packet to agent process **850**, i.e. the process to be conferenced, as illustrated by transmission **1**. Agent process **850** then transmits a CALLACK packet to agent process **830**, as illustrated by transmission **2**, followed by an ANSWER packet, as illustrated by transmission **3**. Agent process **830** then transmits a CONFADD packet, as described hereinafter, to agent process **850**, as illustrated by transmission **4**, and a CONFADD packet to caller process **820**, as illustrated by transmission **5**. The CONFADD packet transmitted to caller process **820** includes the Internet protocol address of agent process **850**, while the CONFADD packet transmitted to agent process **850** contains the IP address of caller process **820**. Each of agent process **850** and caller process **820** then transmit a CONFADDACK packet back to agent process **830** to acknowledge receipt of the conference information. At this point, any AUDIO, VIDEO and/or TEXT packets to be transmitted from a party in the conference will be transmitted to the other parties in the conference. The exchange of packets among the parties, illustrated as AUDIO packets for exemplary purposes only, is represented by transmissions **8–13**. Finally, one of the parties to the conference transmits either an END packet or a CNFDROP packet, as illustrated by transmission **14**, terminating the conference. Note that not all parties to a conference will be disconnected by transmission of this packet. Utilizing this packet sequence and the conferencing functionality in the WebPhone client product, more than one agent may participate in call processing to facilitate quality control, special problem handling, etc.

ACD Thread Processing

Referring to FIG. **9**, a flow chart illustrating the basic process step performed by an initialization thread executing within the ACD server **300** of the present invention is provided. The coding of the process steps of the flow chart of FIG. **9** into instructions or objects having analogous methods suitable for execution by the processor on which the invention is implemented will be understood by those having ordinary skill in the art of programming. Upon initialization of ACD server **300**, a main thread performs initialization tasks similar to those described previously with reference to FIG. **4**, as illustrated by procedural step **900**. Next, an Agent Information Table **330** containing information about available agents is constructed in memory, as illustrated by procedural step **902**. The agent table is constructed from information contained within database **312**. Each entry of the Agent Information Table **330** includes information such as login, password, name, E-mail, split assignment, etc. The main thread next constructs an empty "customer split" structure, i.e. a priority queue, in accordance with a user specified queue depth and in accordance with a group split configuration as defined by the system supervisor, as illustrated by procedural step **904**. The main thread then creates one or more listener threads to listen for socket connect events at the ACD_PORT and the CALL_PORT, as illustrated by procedural step **906**. If a socket event is detected at either the ACD_PORT or CALL_PORT, as illustrated by decisional step **908**, a process thread is created and dispatched to process the socket event, as illustrated by procedural steps **910** and **912**. As described with reference to FIGS. **10-11**, the process thread may comprise a control center, agent or customer thread. For example, if a socket event occurred from the control center, a control center thread is created and dispatched to process incoming requests from the ACD_PORT. Similarly, if a socket event occurred from an incoming caller, a customer thread is created and dispatched to process the customer events from the CALL_PORT.

Once a determination is made that the socket has closed, as illustrated by decisional step **914**, the process thread created to service the request performs any final system maintenance tasks, and terminates, as illustrated by procedural step **916**. The listener thread created in step **906** continues to listen to the appropriate port for new socket events.

Access to the Active Agent Table **340** and the Queue **360** is only allowed with the control of a semaphore to ensure that the information in such tables is not modified while one or more threads have access thereto.

Referring to FIG. **10**, a flow chart illustrating the process steps in accordance with the present invention when the process thread control in procedural steps **910** and **912** is a control center or agent thread. The coding of the process steps of the illustrated flow chart in FIG. **10** into instructions suitable for control of the processor on which the invention

is implemented or into objects having analogous methods for performing the same will be understood by one having ordinary skill in the art of programming. As described previously with reference to procedural step 910 of FIG. 9, a control center or agent thread may be created by a listener thread upon receipt of a socket event at the ACD_PORT. Specifically, an agent thread is created by listener thread, as illustrated by procedural step 1000. The agent thread waits for an ONLINE packet to be received at the socket, as illustrated by decisional step 1010. If the ONLINE packet is received, the agent thread saves the Internet Protocol address of the agent in the Active Agents Table 340, previously described, as illustrated by process step 1012. Next, the agent thread determines if a LOGON packet is received, as illustrated by decisional step 1014. If the LOGON packet is received, the agent entry in the Active Agent Table 340 is updated with a pointer to the agent entry from the Agent Information Table 330, as illustrated by procedural step 1016. Next, using the appropriate entry from the Agent Split Table 350, the thread determines the split to which the agent is assigned and places a pointer in the Active Agent Table 340 to the entry for the subject agent, as illustrated by procedural step 1018. Next, the agent thread send an ON LOINE packet to the connection server on behalf of the agent process, as illustrated by procedural step 1020. Next, the agent thread monitors the connection for status packets indicating the status of the agent, as illustrated by procedural step 1022. Specifically, in the illustrative embodiment, an agent may have the following states. An agent may transmit a "ready" status packet indicating that the agent can now be assigned to an Active Agent Table and can handle customer calls. The agent may transmit a "connected" status packet indicating that the agent actually made contact with a customer. The agent may transmit a "Non-ACD" status packet indicating that the agent has initiated a call and is unavailable to receive customer calls. A "Wrap-Up" packet indicates that the agent has completed a customer call and is currently completing follow-on tasks. Other status packets indicating the non-availability of an agent may be transmitted to indicate that the agent is not currently able to accept customer calls.

If the Agent Information Table 330 indicates that an agent is allowed to originate or terminate non-ACD calls, an ONLINE packet is transmitted to the connection server 252 by the ACD server 300 on behalf of the agent, as illustrated by procedural step 1022. The agent thread monitors, times and logs any instance where call status packets are received, as illustrated by procedural step 1024. If the agent thread determines that either a LOGOFF packet or an OFFLINE packet is received, or, if a socket event is received indicating that the connection was lost, as illustrated by decisional step 1026, the agent thread resets the relevant entry in the Active Agent Table 340, as illustrated by procedural step 1028, and performs any system maintenance before terminating, as illustrated by procedural step 1030.

The process thread created in procedural step 910 of FIG. 9 may also be a customer thread, as described with reference to FIG. 11. Specifically, FIG. 11 is a flow chart illustrating the basic process steps performed by a customer thread in accordance with the present invention. The coding of the process steps illustrated in the flow chart of FIG. 11 into instructions or objects having analogous methods suitable to control the processor on which the invention is implemented will be understood by one having ordinary skill in the art of programming. Following creation of a "customer" thread, as illustrated by procedural step 1100, the customer thread determines whether a CALL packet has been received, as illustrated by decisional step 1102. If the call packet specifies a group or destination not configured to accept calls, the call is logged as a bad call, as illustrated by procedural step 1106. Otherwise, if the group is valid, the customer thread determines the split and agent status for the call, as illustrated by procedural step 1108. Next, the customer thread creates an entry in one of the split queues to log information about the call, as illustrated by procedural step 1110. If the customer thread determines that the split queue has agents available, the thread will transmit an "Agent available" message, as illustrated by decisional step 1112 and procedural step 1114. If the customer thread determines that there are no agents available, as illustrated by decisional step 1112, but that an entry is available in the split queue as illustrated by decisional step 1116, an "All agents are busy" message will be transmitted to the caller process, as illustrated by procedural step 1118. Alternatively, if no entries are available within the split queue, a "Try again later" message is transmitted to the caller process, as illustrated by procedural step 1120. If in decisional step 1112, an agent is available, or, when a queued call receives a "Next available agent" message, the customer thread marks the agent as unavailable, as illustrated by procedural step 1122, and attempts to transfer the call to an agent by transmission of a TRANSFERREQ packet from ACD server 300 to the agent, as illustrated by procedural step 1124. If the transfer request is successful, as determined in decisional step 1126, the customer thread marks the Active Agent Table 340 entry as having a "connected" status, as illustrated by procedural step 1128. Alternatively, if the transfer request was unsuccessful, the customer thread marks the entry within the Active Agent Table 340 to indicate that the agent is temporarily unavailable, as illustrated by procedural step 1130. As a result, ACD server 300 may, optionally, not attempt to transfer another call to the transferred agent until a predetermined time out period has expired, as defined by the ACD user/supervisor. If an agent is unavailable to accept a call, the customer thread attempts to recover the transfer by transferring the call to another agent with a similar TRANSFERREQ packet or, alternatively, leaving the customer in his/her current active queue position, as illustrated by procedural step 1132.

The transmitting of messages as described with reference to the processes illustrated in FIGS. 9–11 is achieved by transferring one or more AUDIO, VIDEO, or other data type packets as previously described with reference to FIGS. 4–8.

ADDITONAL ACD FUNCTIONS

ACD server 300 may be implemented to provide support for the following features:

Abandoned Call Tracking—Abandoned calls or hangups are not transferred from the queues to agents. However, call waiting time till hangup statistics may be kept. In addition, caller information is recorded allowing the ACD server 300 to route the hangups to agents for callback.

Alternate Night CCVs—Alternate call control vectoring based on night mode (see also Time of Day/Week CCV).

Management Defined Agent Availability Modes—Allows call center management to define Agent availability modes. Allows an Agent to report their station status as available to take calls, currently on a call, work mode, break, lunch, personal call, send a fax, sending diskettes, or other management defined status which is reconfigurable dynamically on a split by split basis.

Agent Personal Queues—Agents handling special topics, callers directed to reach a specific agent, emergency calls, supervisors needing to talk to a specific agent can be configured to call the agent specific queue. Agent specific queues are managed, configured and are identical to split queues.

Agent Logon/Log Off—Agent user information is not needed at the agent WebPhone client. Agents login/logout using an agent id and password. Agent information can be looked up by ACD server **300** and reflected up to connection server **252**, if configured to do so by the supervisor. Reflecting this information up to connection server **252** allows the agent to receive and make calls, but allows ACD server **300** to perform call tracking and statistics reporting on these calls.

Agent Tally—The ACD allows agents to enter tally information if configured to do so. The Agent speed dials or uses a directory entry to reach a specific queue, and is prompted to enter the tally information through the Agent WebPhone keypad followed by the END key.

Agent Work Modes—The ACD allows the Agent to specify the progression of work modes in resolving customer calls. In most cases, only some additional paper work or updates to a customer record are necessary. In less frequent cases, the Agent may be required to perform further research or investigation to resolve a problem. These may be billed at different rates. The Agent may enter work mode 'one' when updating the customer data base, work mode 'two' when searching knowledge bases to investigating problems, work mode 'three' when its necessary to duplicate the customer environment, and actually debug a problem as in the case of software products.

Announcements—The ACD can be programmed to replay specific recordings at regular intervals, or greetings when a call is first received.

Assistance—Allows an Agent to request supervisor assistance. This may invoke the programming of the speed dial keys or the dragging of or double clicking on a directory entry for the supervisor queue or a specific supervisor. The supervisor can be privately consulted while the caller is on hold, conferenced in, or the caller transferred.

Automatic Call Recovery—Calls routed to an Agent WebPhone process and not answered within a configured amount of time or number of rings can be re-routed to another available agent.

Automatic Work Mode—Allows the supervisor to configure the ACD server **300** to automatically place the Agent in "available for calls status" after allowing a predetermined time to expire.

Bad Call Reporting—An Agent can report the call as a bad call or mis-routed call.

Break Modes—Allows the Agent to specify that they are unavailable to receive calls for various reasons as configured by the supervisor.

Call Control Vectoring (CCV) on Split Basis—Call Control Vectoring allows the user to configure their ACD to route calls to various queues or splits. Each split may be configured with its own call routing algorithms.

CCV based on Queue Depth—Support for CCV based on queue statistics. Queues reaching certain levels of activity or call hold time to Agent on duty ratios result in alternate CCV to other backup call centers, announcements, or backup/alternate agent strategies.

CCV based on Assigned Agent Priority—assigned to a particular split can be additionally assigned a priority that influences CCV. A collection of Agents assigned the same priority are routed calls equally. As split depths increase, lower priority agents from other splits, working at remote sites, working from home, etc. can be routed calls as needed to satisfy customer needs.

Call Distribution Methods—Configuration options allow the selection of call queuing models based on length of hold time, type of support required (complexity of subject matter, customers paid level of support, etc.)

Call Monitoring (Supervisor or Agent Initiated)—A supervisor may wish to monitor calls to appraise an Agent or insure quality of service. Additionally, supervisors may wish to participate or offer assistance to and Agent.

Call Transfer to Splits—Agents may transfer calls to other queues. Utilized when calls are mis-routed, customer requires attention from multiple agents, etc.

Call Waiting, Queue Depth, and Current Call Statistics—Agents are always aware of the current call load. Additionally, the stations can display the length of the current call, number of agents currently assigned, number of calls currently in the queue, longest call duration on in queue, etc.

Calling Party Identification (CLID, ANI, DNIS, etc.)—Agents receive indication of any of the available information about the incoming call. CLID—Caller ID, ANI—automatic number identification, DNIS—from the CO, or other information collected by voice response units.

Do Not Disturb (DND) and Category Support—Agents may indicate to the ACD that they are currently not available to accept calls for up to 10 supervisor configured classifications. The ACD keeps the statistics for later report generation. For example, Agents may log that restroom facilities are being utilized, on break, at lunch, originating a call to a customer, following up on previous calls requiring special attention, meeting with a supervisor, etc.

Hot Splits—CCV supports routing to Automatic Agents that collect user information or user requirements automatically or just make announcements.

InfoLinks—InfoLinks allow caller information (CLID, DNIS, ANI, etc.) to be passed to other Agent software applications. In addition, Agent WebPhones can utilize Dynamic Data Exchange or Object Linking and Embedding standards.

Language Support/Call Routing—CCV support for call routing based on language requirements.

Maximum Work Mode Time Limit—In automatic work mode, Agents that have not indicated that their work mode is complete can automatically be forced out of work mode, and assigned calls as needed.

Multi-Split Agents—The ACD allows Agents to be assigned to multiple queues by supervisors. In addition, Agents can be reassigned during special peak periods when queue depth reaches excessive levels. See also Queue Based CCV support.

Non-ACD Call Tracking—Support for tracking Agent calls (either business or personal in nature) can be tracked, logged and statistics reported. This applies to both Agent originated or terminated calls.

Priority Queuing—Calls transferred by an Agent to other Agents can receive priority queuing, avoiding being placed back at the beginning of a queue. Calls that have been mis-routed, or calls requiring interaction with multiple agents can be processed in this manner.

Split Features—Separate CCV on a split basis.

Stranded Call Routing—Calls left on the queue for excessive lengths of time can be automatically routed to receive announcements or special Agents.

Supervisor Queues—Support for multiple supervisor queues allowing Agents needing assistance access to the next available supervisor.

Time of Day and Week CCV—Alternate CCV based on the time of day or day of week to support automatic load

balancing. A call center receiving calls after 5:00 pm EST could be route automatically to PST call centers.

ACD Packet Descriptions

The present invention contemplates a number of specialized packets to facilitate interaction between ACD server **300** and agent WebPhone processes **244** and control center **280**, as described hereafter. The AGENT WEBPHONES STATUS CHANGE packet indicates a change in agent WebPhone call state. Through this packet, an agent can report their progress with call processing and indicate availability to accept additional calls. It is desirable that call progress messages and accompanying screens be configurable at ACD server **300** by the supervisor on a split basis, i.e. by call queue.

The AGENT WEBPHONE PARAMETER DOWNLOAD packet allows the ACD supervisor to control some of the configuration parameters of the Agent WebPhone processes, thereby allowing queue configuration changes to be reflected at the agent's desktop.

The LAUNCH packet may be transmitted from control center **280** to ACD server **300** or from control center **280** to connection server **242**. When control center **280** is initiated, the LAUNCH packet is transmitted to ACD sever **300** and connection server **242** thereby creating a persistent socket connection to ACD server **300** and connection server **242**. Control center **280** initially displays the unassigned tab, so the server(s) must supply current unassigned status and keep supplying unassigned events.

SHORT TERM STATISTICS Packet

(Control Center to ACD)

The SHORT TERM STATISTICS Packet may be transmitted from control center **350** to ACD server **300 300** to obtain information about usage over last few hours (# hours desired sent in initiating packet). Examples of this type if information follow:

Total Idle Time per Dept.

Total Processing Time per Dept.

Total DND Time per Dept.

Total Off-line Time per Dept.

CURRENT FULL STATUS Packet

(Control Center to ACD) (Control Center to CS)

Control Center will supply the department desired in the initiating packet. The server should immediately supply current status of the department being viewed and subsequently send all events that happen within this department until the Change View Packet is sent. In the case of the CS, the department is ALL Business s or the unassigned area. In the case of the ACD, the department is either truly a department or the unassigned users area.

CHANGE VIEW Packet

(Control Center to ACD) (Control Center to CS)

This packet will indicate that the Control Center User has changed departments. The server should stop sending event packets for the previous department and send a Full Status Packet for the new department. In the case of the CS, the department is ALL Business s or the unassigned area. In the case of the ACD, the department is either truly a department or the unassigned users area.

EVENT Packet (ACD to Control Center) (CS to Control Center)

This will indicate to the Client that one of the aforementioned events has occurred. The Control Center should update the GUI accordingly.

MOVE USER Packet

(Control Center to ACD) (Control Center to CS)

This packet is used when moving an Agent or Business from one area to another.

The user can be moved to/from all of the following:

Unassigned

Department

General BWP

The ACD/CS is then responsible for making appropriate table changes and subsequently routing calls accordingly.

REMOVE USER Packet

(Control Center to ACD) (Control Center to CS)

The REMOVE USER packet informs the control center **350** to ACD server **300 300** to remove a user from tables or other mechanism to ensure that the user cannot place calls or show up in directory assistance.

ADD DEPARTMENT Packet

Control Center to ACD)

Control Center will ask the ACD to create a new Department.

DELETE DEPARTMENT Packet

(Control Center to ACD)

Control Center will ask the ACD to remove a Department. All s assigned to that department that are not assigned to another department will be moved to the Unassigned areas.

REPORTING Packets

(Control Center to ACD) (Control Center to CS)

Control Center will ask ACD or CS for verbose statistics for printout. Examples of information requested follows:

Call statistics for department in time range

Call statistics for agent in time range

Max/Min call statistics reporting

A software implementation of the above-described embodiments may comprise a series of computer instructions either fixed on a tangible medium, such as a computer readable media, e.g. diskette **142**, CD-ROM **147**, ROM **115**, or fixed disk **152** of FIG. 1A, or transmittable to a computer system, via a modem or other interface device, such as communications adapter **190** connected to the network **195** over a medium **191**. Medium **191** can be either a tangible medium, including but not limited to optical or analog communications lines, or may be implemented with wireless techniques, including but not limited to microwave, infrared or other transmission techniques. The series of computer instructions embodies all or part of the functionality previously described herein with respect to the invention. Those skilled in the art will appreciate that such computer instructions can be written in a number of programming languages for use with many computer architectures or operating systems. Further, such instructions may be stored using any memory technology, present or future, including, but not limited to, semiconductor, magnetic, optical or other memory devices, or transmitted using any communications technology, present or future, including but not limited to optical, infrared, microwave, or other transmission technologies. It is contemplated that such a computer program product may be distributed as a removable media with accompanying printed or electronic documentation, e.g., shrink wrapped software, preloaded with a computer system, e.g., on system ROM or fixed disk, or distributed from a server or electronic bulletin board over a network, e.g., the Internet or World Wide Web.

Although various exemplary embodiments of the invention have been disclosed, it will be apparent to those skilled in the art that various changes and modifications can be made which will achieve some of the advantages of the invention without departing from the spirit and scope of the invention. Further, many of the system components described herein such as the client application and the gateway have been described using products from NetSpeak Corporation. It will be obvious to those reasonably skilled in

the art that other components performing the same functions may be suitably substituted. Further, the methods of the invention may be achieved in either all software implementations, using the appropriate processor instructions, or in hybrid implementations which utilize a combination of hardware logic and software logic to achieve the same results. Such modifications to the inventive concept are intended to be covered by the appended claims.

Call Packet

Packet Name: Call
Packet Type: WPP_CALL
Direction: WebPhone client to Conference Server
Description: Initiate a point to point call or a conference call.

Structure:	Structure Name	Data Type		Description
	Packet Type	unsigned char	1 byte	WPP message identifier
	dwSession	unsigned long	4 bytes	Session ID
	capability	unsigned short	2 bytes	Version Capability
	protocol	unsigned short	2 bytes	Version Protocol
	vendor	unsigned short	2 bytes	Version Vendor
	nCodec	unsigned short	2 bytes	Codec number
	szFirstName	char (10)	10 bytes	first name
	szLastName	char (25)	25 bytes	last name
	szAlias	char (20)	20 bytes	alias
	szEmailAddr	char (90)	90 bytes	EMail
	szIpAddr	char (80)	80 bytes	IP
	szStreetAddr	char (50)	50 bytes	street
	szApt	char (20)	20 bytes	apt
	szCity	char (20)	20 bytes	city
	szState	char (20)	20 bytes	state
	szCountry	char (20)	20 bytes	country
	szZipCode	char (20)	20 bytes	zip code
	szPhone	char (25)	25 bytes	phone number
	szFax	char (25)	25 bytes	fax number
	szCompany	char (25)	25 bytes	company name
	wTime	unsigned short	2 bytes	current time
	cType	char (1)	1 byte	caller type
	dwDecode Key	unsigned long	4 bytes	decode key
	AUDIOCODECS[5]	AUDIOCODEC	25 bytes	audio codec info
	dwFlag	unsigned long	4 bytes	call flag

cType	0 - individual
	1 - company
dwFlag	1 - conference server preferred

0.1. CallAck Packet

Packet Name: Caller Acknowledgment
Packet Type: WPP_CALLERACK
Direction: From the Connection Server

Description: Send Call OK verification.

Structure:	Structure Name	Data Type		Description
	Packet Type	unsigned char	1 byte	WPP message identifier
	dwSession	unsigned long	4 bytes	Session ID or conference ID
	capability	unsigned short	2 bytes	Version Capability
	protocol	unsigned short	2 bytes	Version Protocol
	vendor	unsigned short	2 bytes	Version Vendor
	nCodec	unsigned short	2 bytes	Codec number
	dwRemoteSession	unsigned long	4 bytes	caller's session ID
	szFirstName	char (10)	10 bytes	first name
	szLastName	char (10)	25 bytes	last name
	szAlias	char (10)	20 bytes	alias
	szEmailAddr	char (10)	90 bytes	EMail
	szIpAddr	char (10)	80 bytes	IP
	szStreetAddr	char (10)	50 bytes	street
	szApt	char (10)	20 bytes	apt
	szCity	char (10)	20 bytes	city
	szState	char (10)	20 bytes	state
	szCountry	char (10)	20 bytes	country
	szZipCode	char (10)	20 bytes	zip code
	szPhone	char (10)	25 bytes	phone number

-continued

Structure:	Structure Name	Data Type		Description
	szFax	char (10)	25 bytes	fax number
	szCompany	char (10)	25 bytes	company name
	wTime	unsigned short	2 bytes	current time
	cType	char (1)	1 bytes	caller type
	dwDecode Key	unsigned long	4 bytes	decode key
	AUDIOCODECS[5]	AUDIOCODEC	25 bytes	audio codec info
	dwFlag	unsigned long	4 bytes	call flag
cType		0 - individual		
		1 - company		
dwFlag		1 - conference server preferred		

CnfAdd Packet

Packet Name: Conference party add

Packet Type: WPP_CONFADD

Direction: From to Conference Server

Description: Used to supply the conferencing server with the necessary information for this party to be added to a conference. A CALL packet should have preceded this packet with the dwFlag set to ADD PARTY.

Structure:	Structure Name	Data Type		Description
	Packet Type	unsigned char	1 byte	WPP message identifier
	dwSession	unsigned long	4 bytes	Session ID
	capability	unsigned short	2 bytes	Version Capability
	protocol	unsigned short	2 bytes	Version Protocol
	vendor	unsigned short	2 bytes	Version Vendor
	szFirstName	char (10)	10 bytes	first name
	szLastName	char (25)	25 bytes	last name
	szAlias	char (20)	20 bytes	alias
	szEmailAddr	char (90)	90 bytes	Email
	dwIP	unsigned long	4 bytes	IP address
	dwFirewallIP1	unsigned long	4 bytes	
	dwFirewallIP2	unsigned long	4 bytes	
	dwFirewallIP3	unsigned long	4 bytes	
	dwFirewallIP4	unsigned long	4 bytes	
	dwFirewallIP5	unsigned long	4 bytes	
	dwFirewallIP6	unsigned long	4 bytes	
	szStreetAddr	char (50)	50 bytes	street
	szApt	char (20)	20 bytes	apt
	szCity	char (20)	20 bytes	city
	szState	char (20)	20 bytes	state
	szCountry	char (20)	20 bytes	country
	szZipCode	char (20)	20 bytes	zip code
	szPhone	char (25)	25 bytes	phone number
	szFax	char (25)	25 bytes	fax number
	szCompany	char (25)	25 bytes	company name
	wTime	unsigned short	2 bytes	current time
	cType	char (1)	1 byte	caller type
	dwEncode Key	unsigned long	4 bytes	encode key
	dwDecode Key	unsigned long	4 bytes	decode key
	AUDIOCODECS[5]	AUDIOCODEC	25 bytes	audio codec info
	dwFlag	unsigned long	4 bytes	Flag
	dwConfID	unsigned long	4 bytes	Conference ID
	szConfPassword	char (10)	10 bytes	Conference password
dwFlag		1 - create conference		
		2 - create and add to conference		
		4 - add to conference		

CnfAddAck Packet

Packet Name: Conference party add ACK

Packet Type: WPP_CONFADDACK

Direction: From Conference Server to WebPhone client

Description:

Structure:	Structure Name	Data Type		Description
	Packet Type	unsigned char	1 byte	WPP message identifier
	dwSession	unsigned long	4 bytes	Conference ID
	capability	unsigned short	2 bytes	Version Capability
	protocol	unsigned short	2 bytes	Version Protocol
	vendor	unsigned short	2 bytes	Version Vendor
	dwFlag	unsigned long	4 bytes	Flag
	dwEncode Key	unsigned long	4 bytes	encode key
	dwDecode Key	unsigned long	4 bytes	decode key

dwFlag
1 - conference created
2 - added to conference
4 - request denied
8 - password required

CnfServerInfo Packet

Packet #: 0
Packet Name: Conference server information
Packet Type: WPP_CONFSRVINFO
Direction: From to
Description: Request information about the preferred conference server from the

Structure:	Structure Name	Data Type		Description
	Packet Type	unsigned char	1 byte	WPP message identifier
	dwSession	unsigned long	4 bytes	Session ID
	capability	unsigned short	2 bytes	Version Capability
	protocol	unsigned short	2 bytes	Version Protocol
	vendor	unsigned short	2 bytes	Version Vendor
	dwFlag	unsigned long	4 bytes	Flag

CnfServerInfoAck Packet

Packet Name: Conference server information ACK.
Packet Type: WPP_CONFSRVINFOACK
Direction: From to
Description: Returned information about the preferred conference server from the

Structure:	Structure Name	Data Type		Description
	Packet Type	unsigned char	1 byte	WPP message identifier
	dwSession	unsigned long	4 bytes	Session ID
	capability	unsigned short	2 bytes	Version Capability
	protocol	unsigned short	2 bytes	Version Protocol
	vendor	unsigned short	2 bytes	Version Vendor
	dwFlag	unsigned long	4 bytes	Flag
	szEmailAddr	char (10)	90 bytes	Conference server EMail
	szIpAddr	char (10)	80 bytes	Conference server IP
	szPassword	char (10)	10 bytes	Conference server password
	dwConnectionSpeed	unsigned long	4 bytes	Connference server connection speed
	AUDIOCODECS[5]	AUDIOCODEC	25 bytes	audio codec info

CnfDrop Packet

Packet Name: Conference party drop
Packet Type: WPP_CONFDROP

Direction: From to Conference
Description: Used to indicate that a party is leaving a conference.

Structure:	Structure Name	Data Type		Description
	Packet Type	unsigned char	1 byte	WPP message identifier
	dwSession	unsigned long	4 bytes	Session ID
	capability	unsigned short	2 bytes	Version Capability
	protocol	unsigned short	2 bytes	Version Protocol
	vendor	unsigned short	2 bytes	Version Vendor

ConfParties Packet

Packet Name: Conference Parties Information
Packet Type: WPP_CONFPARTIES
Direction: From the Conferencing Server
Description: Send list of parties in conference in NULL terminated form(strings). The party data continues until all of the data required for the number of parties has arrived. This may mean multiple packets. The party data is as follows. The given lengths are maximums with character strings sent NULL terminated.

Structure:	Structure Name	Data Type		Description
	Packet Type	unsigned char	1 byte	WPP message identifier
	dwSession	unsigned long	4 bytes	Session ID
	capability	unsigned short	2 bytes	Version Capability
	protocol	unsigned short	2 bytes	Version Protocol
	vendor	unsigned short	2 bytes	Version Vendor
	dwFlag	unsigned long	4 bytes	Flag 1 - Add 2 - Drop
	nParties	unsigned short	2 bytes	Number of parties
	wPartyDataLength	unsigned short	2 bytes	Party data length in bytes
	PartyData	BYTE	...	Party data
	dwRemoteSession	unsigned long	4 bytes	caller's session ID
	wTime	unsigned short	2 bytes	current time
	dwDecode Key	unsigned long	4 bytes	decode key
	cType	char (1)	1 bytes	caller type
	szFirstName	char (10)	10 bytes	first name
	szLastName	char (10)	25 bytes	last name
	szAlias	char (10)	20 bytes	alias
	szEmailAddr	char (10)	90 bytes	EMail
	szIpAddr	char (10)	80 bytes	IP
	szStreetAddr	char (10)	50 bytes	street
	szApt	char (10)	20 bytes	apt
	szCity	char (10)	20 bytes	city
	szState	char (10)	20 bytes	state
	szCountry	char (10)	20 bytes	country
	szZipCode	char (10)	20 bytes	zip code
	szPhone	char (10)	25 bytes	phone number
	szFax	char (10)	25 bytes	fax number
	szCompany	char (10)	25 bytes	company name

Transfer Packet

Packet Name: Call Transfer.
Packet Type: WPP_TRANSFER
Direction: From to

Description: Transfer current call to new party.

Structure:	Structure Name	Data Type		Description
	Packet Type	unsigned char	1 byte	WPP message identifier
	dwSession	unsigned long	4 bytes	Session ID
	capability	unsigned short	2 bytes	Version Capability
	protocol	unsigned short	2 bytes	Version Protocol
	vendor	unsigned short	2 bytes	Version Vendor
	szEmailAddr	char (10)	90 bytes	EMail
	szIpAddr	char (10)	80 bytes	IP
	szFirstName	char (10)	10 bytes	first name
	szLastName	char (10)	25 bytes	last name
	szAlias	char (10)	20 bytes	alias
	dwConfID	unsigned long	4 bytes	Conference ID
	szConfPassword	char (10)	10 bytes	Conference password
	dwFlag	unsigned long	4 bytes	transfer flag

dwFlag
1 - Add to Conference
2 - Return to me if no answer
4 - Return to me if answer machine

What is claimed is:

1. In an automatic call distribution system, a method of distributing incoming communications over a packet-switched data network, the method comprising the steps of:

A. determining the online status of at least one agent process;

B. defining at least one queue into which incoming communications over the packet-switched network may be placed, each incoming communication containing user information identifying the process from which the communication originated;

C. selectively associating agent processes with the queue in accordance with predetermined criteria;

D. selectively assigning incoming communications to one of the queues in accordance with predetermined criteria;

E. selectively transferring an incoming communication from a queue to one of the agent processes associated with the queue.

2. A computer program product for use with a computer system, the computer system operatively coupled to a computer network and capable of communicating with one or more processes over the network, the computer program product comprising a computer usable medium having program code embodied in the medium for distributing communications to one or more agent processes, the program code comprising:

A. program code means configured to determine the presence of at least one process operatively coupled to the computer system;

B. program code for defining within the computer system memory a queue, the queue having a plurality of entries, each capable of retaining information associated with an incoming communication;

C. program code, responsive to the agent processes currently online for enabling association of agent processes with the queue in accordance with a predetermined criteria;

D. program code, responsive to incoming communications to the computer system for selectively associating an incoming communication with the queue in memory; and

E. program code, responsive to the incoming communications retained in queue and the association of agent processes with the queue, for selectively transferring an incoming communication to an agent process associated with the queue in which the incoming communication user information resides.

3. An automatic call distribution system for use with a packet-switched data network comprising:

A. a plurality of agent processes operatively coupled to the network

B. an automatic call distribution server operatively coupled to the network, the automatic call distribution server maintaining in a memory thereof a list containing information associated with selected of the agent processes and a list containing information associated with incoming communications; and

C. a control center process operatively coupled to the automatic call distribution server, the control center process further comprising a graphic user interface for visually displaying and modifying the information within lists maintained in the automatic call distribution server memory.

* * * * *

EXHIBITS 11-14

REDACTED IN THEIR ENTIRETY

EXHIBIT 15

US006041114A

United States Patent [19]

Chestnut

[11] **Patent Number: 6,041,114**

[45] **Date of Patent: Mar. 21, 2000**

[54] **TELECOMMUTE SERVER**

[75] Inventor: **Kevin L. Chestnut**, Seattle, Wash.

[73] Assignee: **Active Voice Corporation**, Seattle, Wash.

[21] Appl. No.: **08/825,206**

[22] Filed: **Mar. 27, 1997**

[51] **Int. Cl.**[7] **H04M 3/42**

[52] **U.S. Cl.** **379/211**; 379/93.02; 379/212; 379/214

[58] **Field of Search** 379/210, 211, 379/219, 220, 93.02, 93.03

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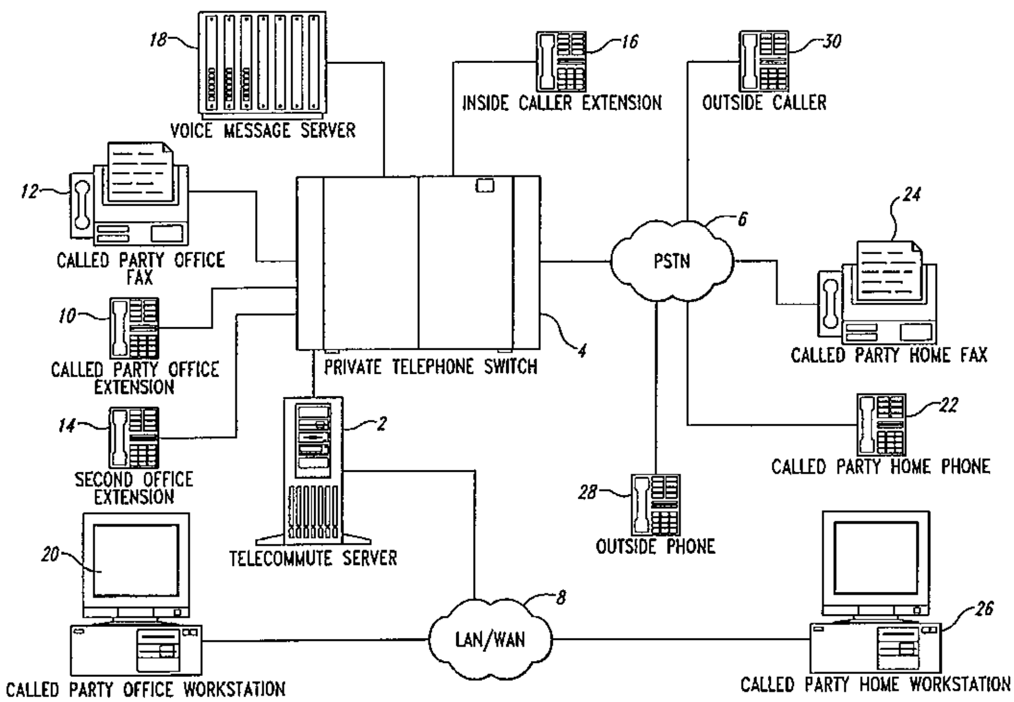
Primary Examiner—Creighton Smith
Attorney, Agent, or Firm—Graybeal Jackson Haley LLP

[57] **ABSTRACT**

A method and device for managing a telecommunication system, including call forwarding, with a computer network (LAN, WAN, etc.) integrated with a private branch exchange (PBX) connected to a Public Switched Telephone Network (PSTN). Calls are forwarded based upon the device used to log onto the computer network by the called party.

41 Claims, 5 Drawing Sheets



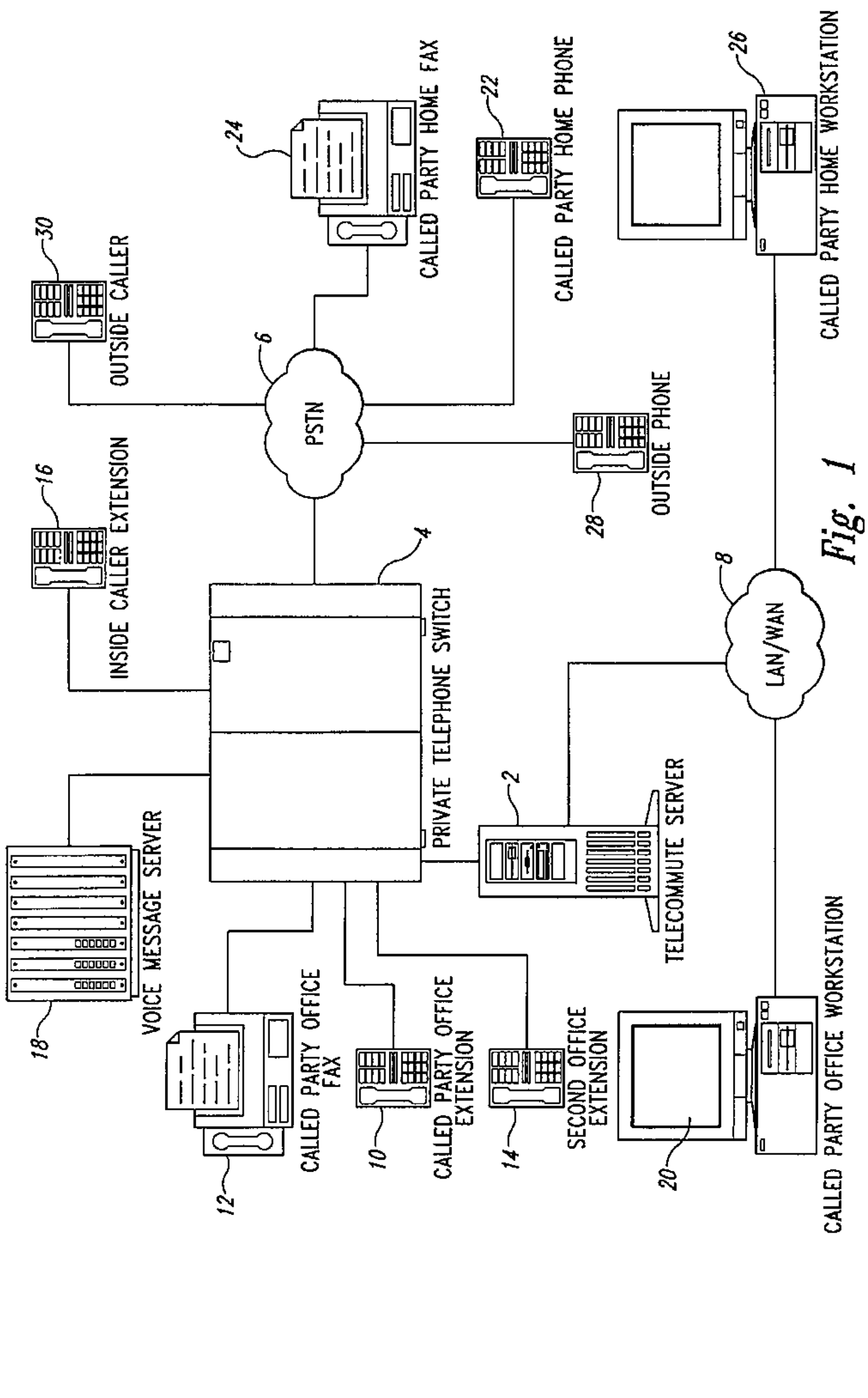

CALLED PARTY HOME FAX
24
CALLED PARTY HOME PHONE
22
CALLED PARTY HOME WORKSTATION
26
OUTSIDE CALLER
30
PSTN
6
OUTSIDE PHONE
28
INSIDE CALLER EXTENSION
16
4
PRIVATE TELEPHONE SWITCH
LAN/WAN
8
2
TELECOMMUTE SERVER
VOICE MESSAGE SERVER
18
CALLED PARTY OFFICE FAX
12
CALLED PARTY OFFICE EXTENSION
10
SECOND OFFICE EXTENSION
14
20
CALLED PARTY OFFICE WORKSTATION

Fig. 1

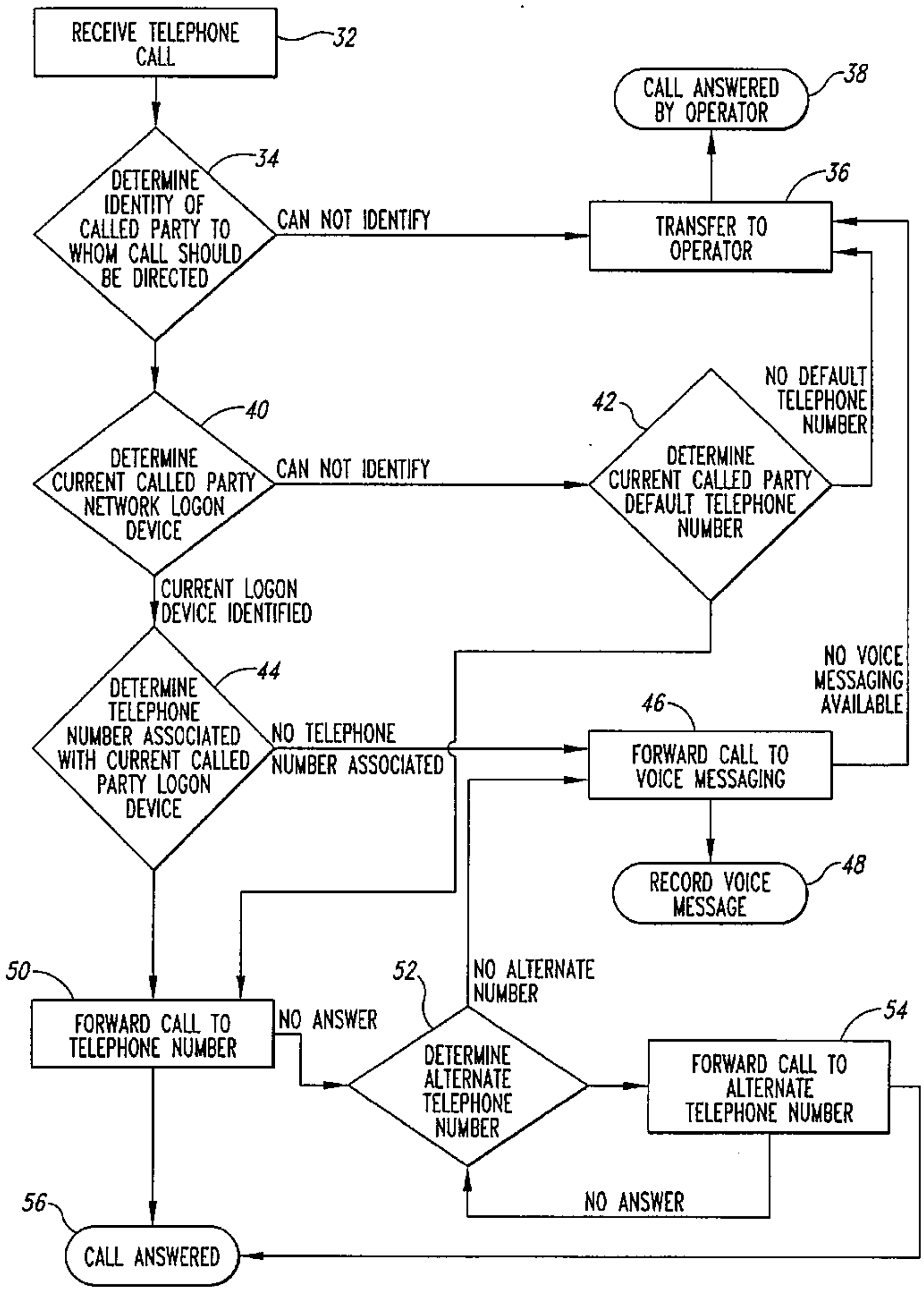
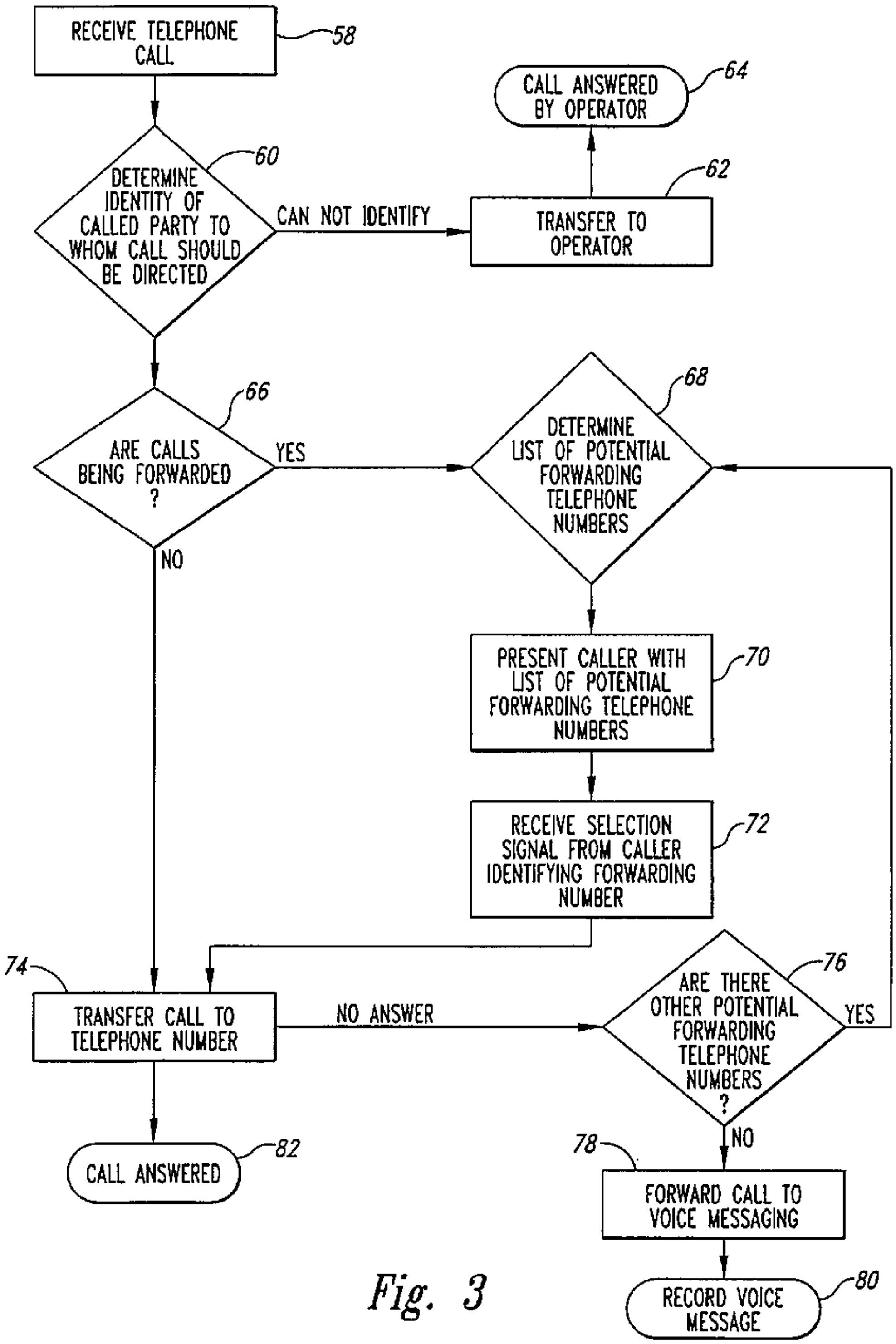

RECEIVE TELEPHONE CALL
32
DETERMINE IDENTITY OF CALLED PARTY TO WHOM CALL SHOULD BE DIRECTED
34
CAN NOT IDENTIFY
TRANSFER TO OPERATOR
36
CALL ANSWERED BY OPERATOR
38
DETERMINE CURRENT CALLED PARTY NETWORK LOGON DEVICE
40
CAN NOT IDENTIFY
DETERMINE CURRENT CALLED PARTY DEFAULT TELEPHONE NUMBER
42
NO DEFAULT TELEPHONE NUMBER
CURRENT LOGON DEVICE IDENTIFIED
DETERMINE TELEPHONE NUMBER ASSOCIATED WITH CURRENT CALLED PARTY LOGON DEVICE
44
NO TELEPHONE NUMBER ASSOCIATED
FORWARD CALL TO VOICE MESSAGING
46
NO VOICE MESSAGING AVAILABLE
RECORD VOICE MESSAGE
48
FORWARD CALL TO TELEPHONE NUMBER
50
NO ANSWER
DETERMINE ALTERNATE TELEPHONE NUMBER
52
NO ALTERNATE NUMBER
FORWARD CALL TO ALTERNATE TELEPHONE NUMBER
54
NO ANSWER
CALL ANSWERED
56

Fig. 2



RECEIVE TELEPHONE CALL
58
CALL ANSWERED BY OPERATOR
64
DETERMINE IDENTITY OF CALLED PARTY TO WHOM CALL SHOULD BE DIRECTED
60
CAN NOT IDENTIFY
TRANSFER TO OPERATOR
62
ARE CALLS BEING FORWARDED ?
66
YES
NO
DETERMINE LIST OF POTENTIAL FORWARDING TELEPHONE NUMBERS
68
PRESENT CALLER WITH LIST OF POTENTIAL FORWARDING TELEPHONE NUMBERS
70
RECEIVE SELECTION SIGNAL FROM CALLER IDENTIFYING FORWARDING NUMBER
72
74
TRANSFER CALL TO TELEPHONE NUMBER
NO ANSWER
ARE THERE OTHER POTENTIAL FORWARDING TELEPHONE NUMBERS ?
76
YES
NO
CALL ANSWERED
82
78
FORWARD CALL TO VOICE MESSAGING
RECORD VOICE MESSAGE
80

Fig. 3

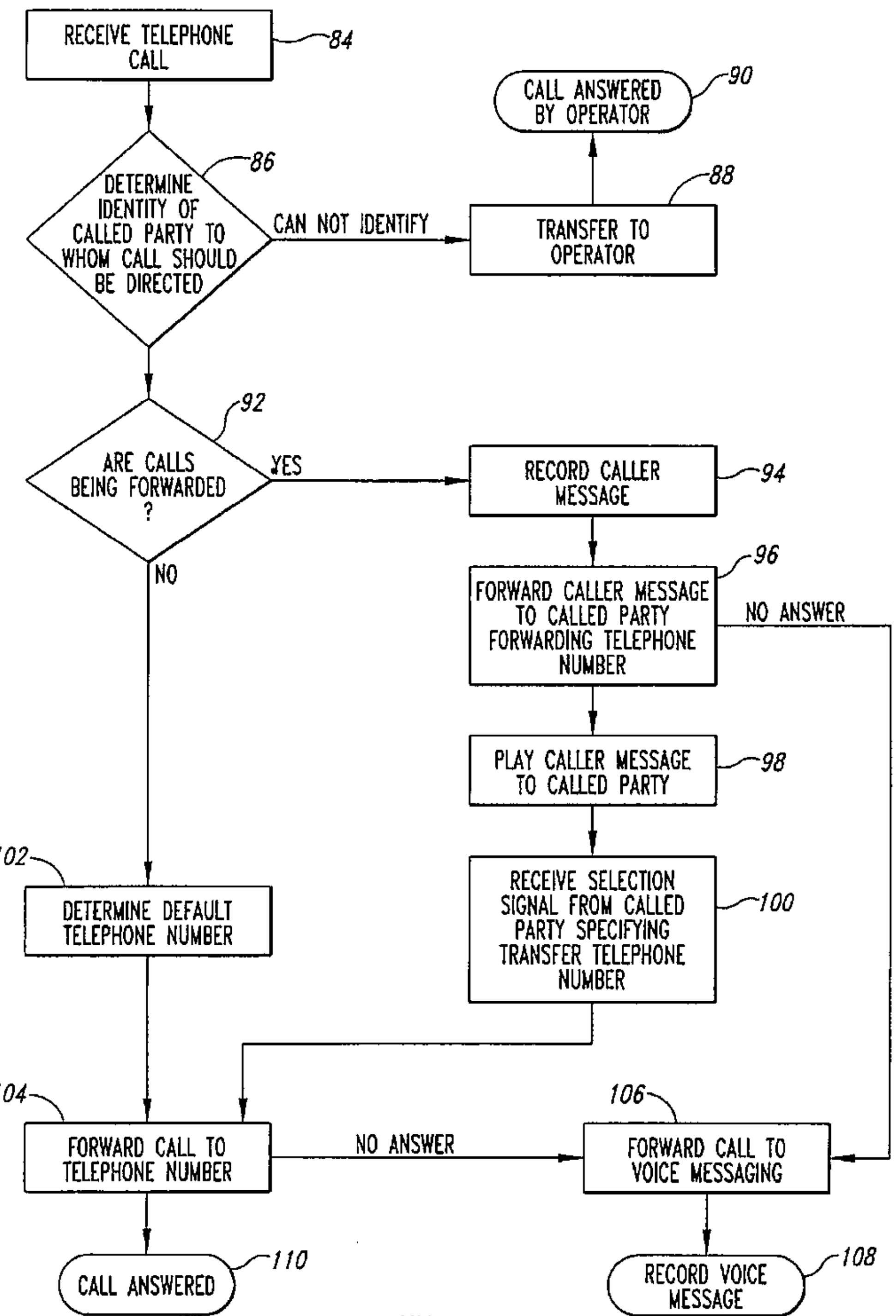
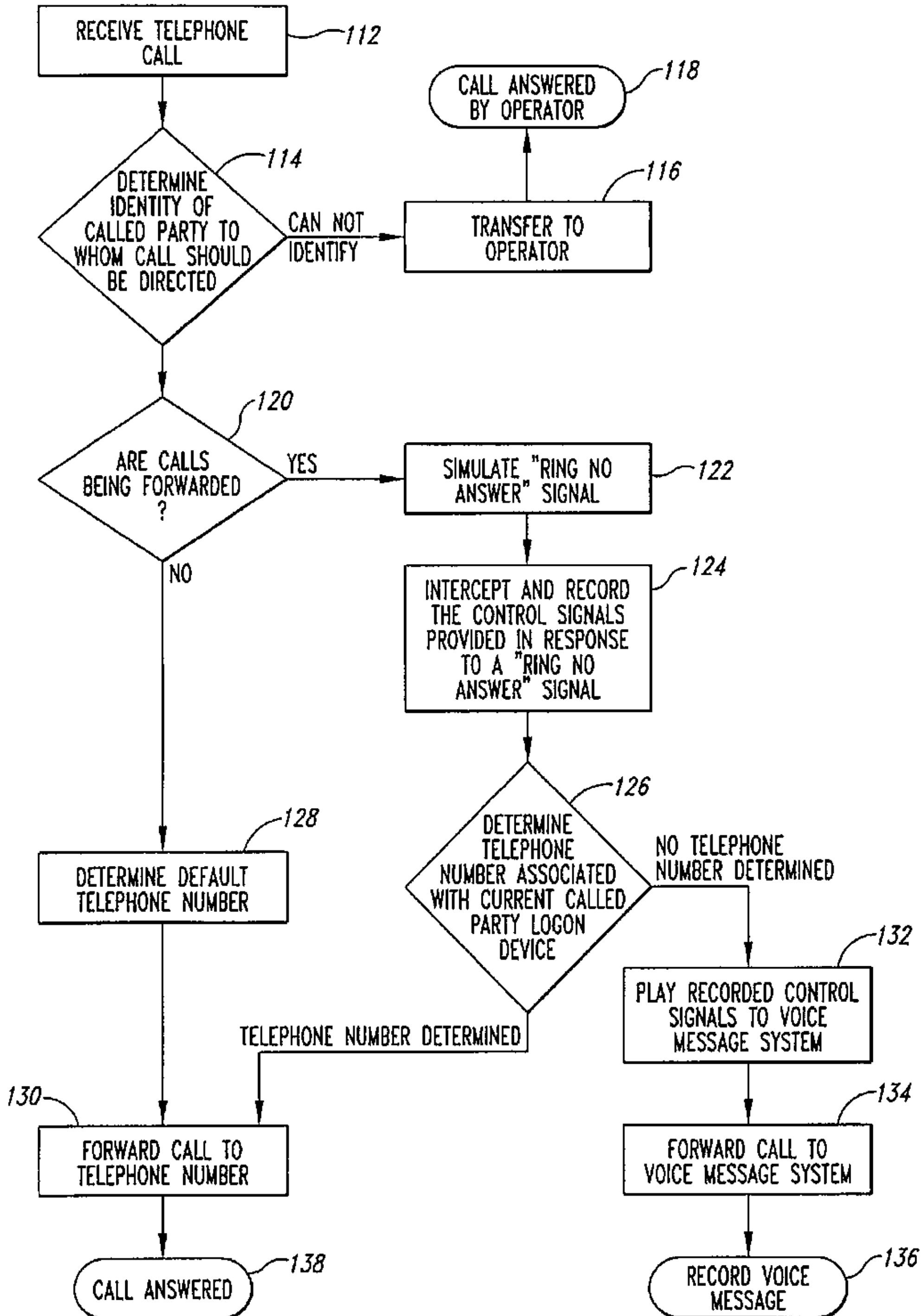

RECEIVE TELEPHONE CALL
84
DETERMINE IDENTITY OF CALLED PARTY TO WHOM CALL SHOULD BE DIRECTED
86
CAN NOT IDENTIFY
TRANSFER TO OPERATOR
88
CALL ANSWERED BY OPERATOR
90
ARE CALLS BEING FORWARDED ?
92
YES
NO
RECORD CALLER MESSAGE
94
FORWARD CALLER MESSAGE TO CALLED PARTY FORWARDING TELEPHONE NUMBER
96
NO ANSWER
PLAY CALLER MESSAGE TO CALLED PARTY
98
RECEIVE SELECTION SIGNAL FROM CALLED PARTY SPECIFYING TRANSFER TELEPHONE NUMBER
100
DETERMINE DEFAULT TELEPHONE NUMBER
102
FORWARD CALL TO TELEPHONE NUMBER
104
NO ANSWER
FORWARD CALL TO VOICE MESSAGING
106
CALL ANSWERED
110
RECORD VOICE MESSAGE
108

Fig. 4



RECEIVE TELEPHONE CALL
112
CALL ANSWERED BY OPERATOR
118
DETERMINE IDENTITY OF CALLED PARTY TO WHOM CALL SHOULD BE DIRECTED
114
CAN NOT IDENTIFY
TRANSFER TO OPERATOR
116
ARE CALLS BEING FORWARDED ?
120
YES
SIMULATE "RING NO ANSWER" SIGNAL
122
NO
INTERCEPT AND RECORD THE CONTROL SIGNALS PROVIDED IN RESPONSE TO A "RING NO ANSWER" SIGNAL
124
DETERMINE TELEPHONE NUMBER ASSOCIATED WITH CURRENT CALLED PARTY LOGON DEVICE
126
NO TELEPHONE NUMBER DETERMINED
DETERMINE DEFAULT TELEPHONE NUMBER
128
PLAY RECORDED CONTROL SIGNALS TO VOICE MESSAGE SYSTEM
132
TELEPHONE NUMBER DETERMINED
130
FORWARD CALL TO TELEPHONE NUMBER
FORWARD CALL TO VOICE MESSAGE SYSTEM
134
CALL ANSWERED
138
RECORD VOICE MESSAGE
136

Fig. 5

TELECOMMUTE SERVER

FIELD OF THE INVENTION

The present invention relates generally to a system for managing a telecommunications system, and more particularly to a telecommunications management system which controls call forwarding based upon user activity on an associated computer terminal.

BACKGROUND OF THE INVENTION

Telecommuting is the substitution of telecommunications technology for the trip to and from the primary workplace. Computers, cellular phones, voice messaging, fax machines, and advanced communications links such as Integrated Services Digital Network (ISDN) and dial-up access have removed the barriers that once required workers to be in their offices. Telecommuting applies to employees working at home, employees working from a satellite office, and employees working "on the road".

The potential advantages of telecommuting are numerous and varied. Beyond the obvious advantages such as reduced rush hour traffic and enhanced air quality, there are a number of less obvious advantages such as increased employee productivity and expanded geographic range. Additionally, total office space requirements can be reduced when employees work at home, satellite offices can be established with lower overhead and are possible in areas that would have been geographically prohibitive, and emergency preparedness is improved through the decentralization of resources.

The Local-Area Network (LAN) is fast becoming the technology backbone of today's offices, since more and more computing and information resources are based on the LAN. Office workers who come to rely on easy LAN access need the same kind of access when they are working away from the office.

While electronic mail grows in popularity, the telephone and accompanying voice messaging systems are still a necessary part of the modern business environment. Computer and telephone systems are being linked through Computer Telephony Integration (CTI) applications which facilitate incoming and outgoing call handling and control.

CTI applications can be used to seamlessly interface the caller, the called party, and information on a host computer for a variety of applications. CTI applications deliver caller ID, automatic number identification (ANI), dialed number identification services (DNIS), and interactive voice response (IVR) dialed digits, such as a customer's account number, to a software application. CTI applications can also deliver request signals, such as "hold call" or "transfer call", to a telephone system.

Numerous prior art systems allow employees to access a Local Area Network via a remote dialup. Once connected they can access most of the resources of the LAN as if they were in the office. However, since the telephone they are using is not part of the office phone system they are cut off from the bulk of the CTI application functions they have available to them at the office. Some systems may allow them to listen to voice mail, however they are no longer able to use any applications which require them to have access to a telephone connected to the office telephone system. Other prior art systems allow employees to remotely access voice messaging and set call forwarding through the use of Dual Tone Multi Frequency (DTMF) tones from a touch tone phone.

In most prior art systems, the lack of integration between a company's telephone system and LAN means that an employee has to call in to the company's phone system to check their voice messaging, manually set call forwarding, and then remotely log on to the company's LAN. After call forwarding is set up, people calling the employee will have lost the ability to leave voice messaging or the employee will have to continue to call in to check their voice messaging. After logging off the LAN, the employee must remember to call into the company's telephone system to discontinue call forwarding. Furthermore, there are numerous telephone systems which do not even afford this level of connectivity, which in turn makes telecommuting a less viable alternative.

In order for a company and its employees to obtain the fullest benefit from telecommuting, communications between telecommuting employees, the primary office, and the outside world must be managed efficiently. The management of telecommunications resources extends to telephone and data communications alike. There is a need for a telecommunications management system which closely integrates a company's LAN with its telephone network and makes the same CTI application functions available to an employee whether they are in the office or working from a remote location.

The present invention closely integrates a company's LAN with its telephone network and controls call forwarding based upon user activity on an associated computer terminal. The present invention extends the functionality of the office telephone system to whatever phone the employee has available at a remote location.

SUMMARY OF THE INVENTION

The present invention, referred to as a telecommute server, is a method for controlling call forwarding using a computer connected to a data network and a telephone network. The call is forwarded based upon whether or not the called party is logged onto the data network. The forwarded call is directed to a telephone line associated with the terminal from which the called party is logged on. The called party may be associated with a particular extension and calls directed to that extension will ring through to the phone associated with the computer the called party is currently logged onto.

Call forwarding is terminated when the called party logs off or the connection is broken. The called party may instruct the system to continue call forwarding for a specified amount of time after a disconnection or they log off. Call forwarding may also be scheduled for a predefined period of time after an initial logon regardless of whether the computer is logged on or off.

Call forwarding based on computer logon may be further scheduled so that calls are forwarded to different telephone lines associated with telephones or voice messaging systems depending upon a predefined schedule. Alternatively, call forwarding may be made conditional based upon other information received by the telephone system, such as caller ID or ANI. The system can also be set up to alter the schedule if it detects that the called party is logged onto a terminal associated with a different telephone extension than the one defined in the schedule.

Logging on to the data network may cause more than one phone line to be forwarded. By way of example, logging on from a computer at home may cause voice phone calls to be forwarded to one telephone line associated with the called party's home and fax calls directed to a particular fax machine to be forwarded to another location. Also, the type

of connection used to log on may serve to identify which extension the calls should be forwarded to.

Calls may originate from outside or from within the company and may be forwarded within the company or to an outside line. This is an important feature of the invention because it makes employees just as accessible as they would be if they were at their desk in the employer's office.

Another aspect of the present invention provides a method for controlling call forwarding by providing the caller with the option of trying the called party at a second location if they are not available at a first location.

In one embodiment, the caller may be provided with a list of locations, any of which can be selected by the caller and tried in order to locate the called party. The list may be modified by the day of the week, the time of day, or whether or not the called party is currently logged on from a remote location. The list may also offer the caller the option to have the call forwarded to a third party.

Additionally, the system may also provide different callers with different levels of access to call forwarding options. Callers may be identified through "caller ID", inputting an identifying code via the telephone touchpad, or some other method of identification. Unknown or low priority callers may only be given the option of leaving a message or having the call transferred to another party while a higher priority caller may be given the option of trying to reach the called party at home.

The system may also be set up to record a message from the caller to be played to a remote called party as part of determining how best to forward the call. The call forwarding options may be automatic or may be presented to the caller or the called party in the form of a menu. The menu may be presented audibly over the phone line or it may be presented in list form on a display. The display may either be part of a communications device or a separate computer display.

The system of the present invention may also be used in conjunction with a Network Switch Server (NSS) which would give the caller the ability to respond to a call forwarding option menu from a computer terminal via a data network.

The present invention also includes a call progress manager which controls the protocols used to forward a call depending upon where the call originated and where it was forwarded to. Progress tones such as busy, trunk busy (reorder), ring no answer, answered by human, answered by machine, are managed. The present invention generates the necessary control signals to respond to the progress tones generated by the outside telephone network.

The system of the present invention can distinguish between internal extensions, outside lines, cell phones, Internet voice, and 2 way pagers. For example, on internal calls when there is "no answer", the system can be instructed to intercept for remote presence determination and ring at remote location while calls from outside the company are sent to a voice messaging system. Remote presence determination includes checking to see if the party being called is logged onto the data network or if they have scheduled to have calls forwarded at this time.

The present invention, a telecommute server, can either be integrated into a system which includes voice messaging or may be used as a stand-alone system which can be connected to a separate voice messaging system. The telecommute server intercepts incoming calls which would be forwarded to voice mail because of a "ring no answer" progress tone, records the DTMF tones which would be provided to the voice messaging system, and checks to see if there is an alternative line to which the call should be forwarded. If there is no alternative line to which the call should be forwarded, the system telecommute server passes the call onto the voice messaging system. If there is a line to which the call should be forwarded, the telecommute server forwards the call to the specified line. If there is no answer at the forwarded number, the Telecommute Server transfers the call back to the voice messaging system and plays the earlier recorded DTMF tones to the voice messaging system. The voice messaging system then answers the call as it would have without the presence of the telecommute server. The telecommute server can, through recording the DTMF tones, control any DTMF controlled device. The system can be implemented so as to work with any prior art device whether it uses in-band or outband signaling.

These and other features of the present invention will be more fully appreciated when considered in light of the following detailed description and drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. 1 is a functional diagram of the present invention.

FIG. 2 is a flowchart of the method of the present invention.

FIG. 3 is a flowchart of the method of the present invention.

FIG. 4 is a flowchart of the method of the present invention.

FIG. 5 is a flowchart of the method of the present invention.

DETAILED DESCRIPTION OF THE INVENTION

FIG. 1 shows the telecommute server 2 connected to a computer network 8 and a private telephone switch (private branch exchange (PBX)) 4 which in turn is connected to a Publicly Switched Telephone Network (PSTN) 6. A called party office extension 10, a called party office fax machine 12, a second office extension 14, an inside caller extension 16, and a voice messaging system 18 are also connected to the PBX 4. A called party office work station 20 is connected to the computer network 8. Called party home phone 22, called party home fax 24, outside phone 28, and outside caller 30 are all connected to PSTN 6. A called party home workstation 26 is connected to the computer network 8.

When an outside caller 30 places a call on the PSTN 6 the call is directed to the called party office extension 10 by the private branch exchange 4. Before the PBX sends the call to the called party office extension 10, the telecommute server 2 checks the computer network 8 to see if the called party is logged on. If the called party is logged on, the telecommute server 2 instructs the private branch exchange 4 to forward the call to the telephone extension associated with the device the called party has used to log onto the computer network 8.

If the called party was logged onto the computer network 8 from the called party office workstation 20, then the call would be directed to the called party office extension 10. If the called party were logged onto the computer network 8 from the called party home workstation 26, then the telecommute server 2 would instruct the PBX 4 to forward the call to called party home phone 22. The telecommute server 2 selects the telephone number to which incoming calls should be forwarded based upon a record stored in a memory which associates a forwarding telephone number, such as the

number for called party home phone 22, with a network logon device, such as called party home workstation 26.

If the connection between the network logon device, called party home workstation 26 or called party office workstation 20, and the computer network 8 is interrupted, intentionally (via a logoff) or accidentally (via a disconnect), the telecommute server 2 can continue to forward calls for a specified period of time after a disconnect or logoff. Alternatively, the telecommute server 2 can continue to forward calls to a previously associated telephone number for a specified period of time after a disconnect but forward calls to another telephone number or a voice message system after the called party logs off. The telecommute server 2 may either have the call forwarding preferences preprogrammed into it or the forwarding preferences may be entered by the called party when he/she logs onto or off of the computer network 8.

The telecommute server 2, can also forward incoming calls based upon other criteria including day or date, time of day, the identity of the caller, or any preprogrammed set of rules. It is within the scope of the invention for the telecommute server 2 to utilize a set of forwarding preferences which are based the above criteria as well as other factors such as who else in the office is logged onto the computer network 8 or the telephone extensions currently in use.

If the called party is not currently logged onto the computer network 8, the telecommute server 2 will instruct the PBX 4 to direct the call to a default telephone number. In most instances, the called party office extension 10 will be the default telephone number. If the called party office extension 10 is not answered (generating a "ring no answer" signal), the PBX 4 may forward the call to a voice messaging system 18. Alternatively, the telecommute server 2 may instruct the PBX 4 to send the incoming call to a voice messaging system 18 if the called party is not logged onto the computer network 8.

In another embodiment of the present invention, the telecommute server 2 will be used with a voice messaging system 18 that requires information, in the form of control signals, from the PSTN 6 or PBX 4. When the telecommute server intercepts an incoming call to check if the called party is logged onto the computer network 8, it also records any control signals that would normally be provided to the voice messaging system from the PBX 4 or PSTN 6. If the telecommute server identifies that the called party is logged on, then it will forward the call to the appropriate telephone number. If the call is forwarded to a telephone number and there is no answer, then the telecommute server 2 plays the appropriate control signals to the voice messaging system 18.

The telecommute server 2 can also be set up to present a caller with a menu listing locations to which the call can be forwarded. The caller then selects a location, most likely using the telephone touchpad, and the telecommute server forwards the call to the selected location. If there is no answer, the telecommute server 2 can either transfer the call to a voice messaging system 18 or try another location. The menu presented to the caller may be modified based upon whether or not the called party is logged onto the computer network 8, time of day, day or date, or the caller's identity.

In another embodiment, the telecommute server 2 can ask the caller to record a message for the called party. The message is then forwarded to and played for the called party. The called party is then presented with a menu which allows him to take the call, record a message to be played for the calling party, transfer the call to a voice messaging system, or transfer the call to another telephone number. The options available to the called party may be modified based upon whether or not the called party is logged onto the computer network 8, time of day, day or date, or the caller's identity.

FIGS. 2–5 illustrate the methods embodied by the present invention. Reference numerals below refers to described steps in the method, not to any noun that they happen to follow.

In FIG. 2 a telephone call is received 32 and the identity of the called party is determined 34. If the called party can not be identified, the call is transferred to an operator 36 and the call is answered by an operator 38.

The identity of the called party is determined 34 by looking up the dialed extension in an index stored in a computer memory and storing the identity of the associated called party stored in a memory. If the identity of the called party is determined, then the next step is to determine the current called party network logon device 40. The current called party network logon device is determined 40 by comparing identity of the called party, which is stored in a memory, with a list of persons currently logged onto the computer network and the network identifier for the device with which they logged on to the computer network.

If no current logon device is identified, then the current called party default telephone number is determined 42 by comparing the identity of the called party, stored in a memory, with a list of default telephone numbers indexed by called party. If no default telephone number is available then the call is transferred to the operator 36 and the call is answered by an operator 38. If a default telephone number is determined 42 then the call is forwarded to the telephone number 50 and the call is answered 56.

If the current called party network logon device is determined then the telephone number associated with the current called party network logon device is determined 44 by comparing the identity of the logon device with a list of telephone numbers indexed by logon device stored in a memory. Other factors including time of day, day of the week, date, and/or the identity of the calling party may be used to determine the forwarding number by providing additional indexing criteria. The call is then forwarded to the identified telephone number 50. If no telephone number is associated with the current logon device, then the call is forwarded to a voice messaging system 46 and a message is recorded 48.

If the forwarded call is not answered, then an alternate forwarding number is determined 52 and the call is forwarded to the alternate telephone number 54. The alternate forwarding number is determined 52 in the same fashion as the telephone number associated with the current called party network logon device is determined 44 and additional factors may apply to the determination of the telephone number to which the call should be forwarded. If there is no answer, then a second alternative forwarding number will be identified 52 and the call is forwarded 54 to the second alternative forwarding number. If there is no alternative forwarding number available, the call is forwarded to a voice messaging system 46 and a message is recorded 48.

In FIG. 3 a telephone call is received 58 and the identity of the called party is determined 60. If the called party can not be identified, the call is transferred to an operator 62 and the call is answered by an operator 64.

If the called party is identified, then the system checks to see if calls are being forwarded 66. If calls are being forwarded, then a list of potential forwarding numbers will be determined 68. The list of potential forwarding numbers

can be based on one or more preprogrammed criteria, including the identity of the called party's current or most recent network logon device, day of the week, date, time of day, and/or the identity of the caller. The caller is then presented with a list of potential forwarding telephone numbers. These numbers may be presented as locations ("home phone, car phone, cell phone") or the caller may be offered options to "try another location or leave a message". As discussed above, different lists may be presented to different callers based on their identity or the source of origin of their call, and the lists of potential forwarding numbers may be effected by the time of day or other criteria. The caller then selects the telephone number (location) they want the call forwarded to. This selection may be made by pressing a key on the telephone keypad or speaking the selection into the receiver or, if the caller is connected via computer-telephone integration, by selecting a screen item with a mouse or pressing a key. The selection signal is received **72** and the call is transferred to the telephone number associated with the selection signal **74**.

If calls are not being forwarded **66**, then the call is transferred **74** to the originally dialed telephone number or the extension to which a PBX had transferred the call.

If there is no answer at the originally dialed telephone number, then the call will be forwarded to voice messaging **78** and a message will be recorded **80**. If there is no answer at a forwarded telephone number, then other potential forwarding numbers will be identified **76**. If there are other potential forwarding numbers, then a second list of potential forwarding numbers will be determined **68** and presented to the caller **70** and the forwarding process will be repeated. If there are no other potential forwarding telephone numbers or calls are not being forwarded, then the call will be forwarded to a voice messaging system **78** and a message recorded **80**.

In FIG. **4** a telephone call is received **84** and the identity of the called party is determined **86**. If the called party can not be identified, the call is transferred to an operator **88** and the call is answered by an operator **90**.

If the called party is identified, then the system checks to see if calls are being forwarded **92**. If calls are being forwarded, then a voice message from the caller is recorded **94**. The caller's message is then forwarded to the called party's forwarding telephone number **96**. If the telephone is answered, the caller's message is played for the called party **98**. A selection signal is received from the called party **100** and the call is transferred to the telephone number associated with the selection signal **104**. In the preferred embodiment, the called party is presented with a list of potential forwarding numbers, including transferring the call to the called party or to a voice messaging system. The list of potential forwarding numbers can be based on one or more preprogrammed criteria including the identity of the called party's current or most recent network logon device, day of the week, date, time of day, the source of origin of the call, and/or the identity of the caller.

If calls are not being forwarded **92**, then the default telephone number is determined **102** and the call is forwarded to the default number **104**. If there is no answer at the called party forwarding number **96** or the telephone number to which a call has been forwarded **104**, then the call is forwarded to a voice messaging system **106** and a message is recorded **108**.

In FIG. **5** a telephone call is received **112** and the identity of the called party is determined **114**. If the called party can not be identified, the call is transferred to an operator **116** and the call is answered by an operator **118**.

If the called party is identified, then the system checks to see if calls are being forwarded **120**. If calls are being forwarded, then a "ring no answer" signal is simulated and played back to the telephone network **122**. The control signals provided by the telephone network in response to the "ring no answer" signal are intercepted and recorded **124**. The signals can be in band DTMF tones, some other in band signalling system, or an out of band signalling system. If out of band tones are used the signalling line must be monitored as well as the communications line. The telephone number associated with the called party's current network logon device is determined **126**, and the call is forwarded to that telephone number **130**. Alternatively the call could be forwarded to a telephone number based upon some criteria other than the called party's current logon location.

If calls are not being forwarded **120**, then the default telephone number is determined **128** and the call is transferred to that number **130**. If there is no answer at that number, the prior art voice message system will record a message as usual.

If no forwarding telephone number is determined **126**, then the recorded control signals are played to the voice message system **132** and the call is transferred to the voice message system **134**. The voice message system responds as if there had been no interruption in the call and records a voice message **136** as if the "ring no answer" control signals had been received directly from the telephone network.

From the foregoing teachings, it can be appreciated by one skilled in the art that a new, novel, and nonobvious telecommunication management system has been disclosed. It is to be understood that numerous alternatives and equivalents will be apparent to those of ordinary skill in the art, given the teachings herein, such that the present invention is not to be limited by the foregoing description but only by the appended claims.

I claim:

1. A method for managing a telecommunications system in which call forwarding is determined by whether a computer terminal is logged into a computer network, comprising:

a) receiving a call on a telephone system which is coupled to a computer network;

b) determining with a server the identity of a called party to whom said call should be directed;

c) identifying with the server one of a plurality of network logon devices associated with said called party that is logged-on to said computer network;

d) identifying with the server a telephone number associated with said logged-on network logon device; and

e) forwarding the call to said telephone number, the forwarded call bypassing the server.

2. The method of claim **1**, wherein said call is directed to a voice messaging system if none of said plurality of network logon devices for the called party is identified as logged-on.

3. The method of claim **1**, wherein said call is directed to a telephone number associated with the previously logged-on called party network logon device if no currently logged-on network logon device is identified.

4. The method of claim **1**, wherein said call may be forwarded to any one of a plurality of telephone numbers and the determination of which telephone number said call is forwarded to is based upon the date and time said call is received.

5. The method of claim **1**, wherein said call may be forwarded to any one of a plurality of telephone numbers

and the determination of which telephone number said call is forwarded to is based upon whether said call originated from the publicly switched telephone network or an internal extension.

6. The method of claim 1, wherein said call may be forwarded to any one of a plurality of telephone numbers and the determination of which telephone number said call is forwarded to is based upon a set of predefined rules.

7. The method of claim 1, wherein said call may be forwarded to any one of a plurality of telephone numbers and the determination of which telephone number said call is forwarded to is based upon incoming signals accompanying the call which signals identify the calling party.

8. The method of claim 1 wherein the forwarding the call to said telephone number comprises forwarding the call via a publicly switched telephone network.

9. A method for managing a telecommunications system in which call forwarding is determined by whether a computer terminal is logged into a computer network, comprising:

a) receiving a call on a telephone system which is coupled to a computer network;

b) determining the identity of a called party to whom said call should be directed;

c) determining whether one of a plurality of network logon devices associated with said called party is logged onto said computer network;

d) if one of the network logon devices is logged onto said computer network, then identifying a telephone number associated with said logged-on network logon device and forwarding the call to said telephone number; and

e) if none of said plurality of network logon devices is logged onto said computer network, then directing the call to a default telephone number.

10. A method for managing a telecommunications system in which call forwarding is determined by whether a computer terminal is logged into a computer network, comprising:

a) receiving a call on a telephone system which is coupled to a computer network;

b) determining the identity of a called party to whom said call should be directed;

c) determining whether one of a plurality of network logon devices associated with said called party is logged onto said computer network;

d) if one of the network logon devices is logged onto said computer network, then identifying a telephone number associated with said logged-on network logon device and forwarding the call to said telephone number; and

e) if none of said plurality of network logon devices is logged onto said computer network, then directing said call to a telephone number associated with a previously logged-on network logon device for a specified period of time after said previously logged-on network logon device logs off said computer network.

11. A method for managing a telecommunications system in which call forwarding is controlled by a calling party, comprising:

a) receiving a call from the calling party on a telephone network requesting communications with a called party;

b) presenting said calling party with a menu listing a plurality of locations to which the call can be forwarded;

c) receiving a selection signal from said calling party identifying the location to which said call is to be forwarded; and

d) forwarding said call to a forwarding telephone number associated with said selection signal.

12. The method of claim 11, wherein said menu listing is modified based upon the identity of the calling party.

13. The method of claim 11, wherein said menu listing is modified based upon the time at which the call is received.

14. The method of claim 11, wherein said menu listing is modified based upon the day and date on which the call is received.

15. The method of claim 11, wherein said menu listing includes the option of leaving a message with a voice mail system.

16. The method of claim 11, further comprising the step of:

forwarding said call to a voice messaging system if there is no answer at the telephone number to which the call was forwarded.

17. A method for managing a telecommunications system in which call forwarding is controlled by the called party, comprising:

a) receiving an indication that calls directed to a first communications device should be forwarded to a second communications device;

b) receiving a call from a calling party on a telephone network directed to said first communications device;

c) recording a calling-party message from said calling party;

d) forwarding said calling-party message to said second communications device;

e) playing said calling-party message at said second communications device;

f) receiving a selection signal from said second communications device indicating a third communications device to which said call is to be forwarded; and

g) forwarding said call to said third communications device.

18. The method of claim 17, wherein said second communications device is selected from a plurality of communications devices based upon the time at which the call is received.

19. The method of claim 17, wherein said second communications device is selected from a plurality of communications devices based upon the day and date on which the call is received.

20. The method of claim 17, wherein said third communications device is an auto attendant system.

21. The method of claim 17, wherein said third communications device is a voice messaging system.

22. The method of claim 17, further comprising the step of:

forwarding said call to a voice messaging system if there is no response from said second communications device.

23. The method of claim 17, further comprising the step of:

presenting at said second communications device a menu listing a plurality of devices to which the call can be forwarded.

24. The method of claim 23, wherein said menu listing is modified based upon the identity of the calling party.

25. The method of claim 23, wherein said menu listing is modified based upon the time at which the call is received.

26. The method of claim 23, wherein said menu listing is modified based upon the day and date on which the call is received.

27. The method of claim 23, wherein said menu listing includes the option of forwarding the call to a voice mail system.

28. A method for managing a telecommunications system, which includes a voice messaging system, in which call forwarding is determined by whether a computer terminal is logged onto a computer network, comprising:

a) receiving a call on a telephone system which is coupled to a computer network;

b) determining the number of a called party's extension to which said call should be directed;

c) simulating a "ring no answer" of said extension number by sending a "ring no answer" signal to the telephone system;

d) intercepting control signals provided by the telephone system to the voice messaging system in response to the "ring no answer" signal;

e) recording the control signals which are provided by the telephone system to the voice messaging system in response to the "ring no answer" signal;

f) identifying which one of a plurality of network logon devices associated with said called party is logged-on to said computer network;

g) if no network logon device for the called party is identified as logged-on, playing the control signals to the voice messaging system, in order to transfer the call to said voice messaging system;

h) if a network logon device is identified as logged-on, identifying a telephone number associated with said network logon device and forwarding the call to said telephone number.

29. The method of claim 28, wherein said call is directed to a default telephone number if none of said plurality of network logon devices for the called party is identified as logged-on.

30. The method of claim 28, wherein said call is directed to a telephone number associated with the previously logged on called party network logon device if no currently logged on network logon device is identified.

31. The method of claim 28, wherein said call is directed to a telephone number associated with the previously logged on called party network logon device for a specified period of time after said network logon device logs off the network if no currently logged-on network logon device is identified.

32. The method of claim 28, wherein said call is forwarded to one of a plurality of telephone numbers based upon the date and time said call is received.

33. The method of claim 28, wherein said call is forwarded to one a plurality of telephone numbers based upon whether said call originated from the publicly switched telephone network.

34. The method of claim 28, wherein said call is forwarded to one of a plurality of telephone numbers based upon a set of predefined rules.

35. The method of claim 28, wherein said call is forwarded to one of a plurality of telephone numbers based upon the identity of the calling party.

36. A server for managing a telecommunications system that includes a computer system having a plurality of network logon devices associated with a called party and that includes a telephone system coupled to a publicly switched telephone network, the server operable to:

a) receive information from the telephone system regarding an incoming call directed to the called party;

b) identify the called party from the information;

c) identify one of the network logon devices that is logged onto the computer network;

d) identify a telephone number associated with the logged-on network logon device; and

e) control the telephone system to forward the call to the telephone number, the forwarded call bypassing the server.

37. A server for managing a telecommunications system that includes a computer system having a plurality of network logon devices associated with a called party and that includes a telephone system, the server operable to:

a) receive information from the telephone system regarding an incoming telephone call directed to the called party;

b) identify the called party from the information;

c) determine whether one of the network logon devices is logged onto the computer network;

d) if one of the network logon devices is logged onto the computer network, then identify a telephone number associated with the logged-on network logon device and control the telephone system to forward the call to the telephone number; and

e) if none of the network logon devices are logged onto the computer network, then control the telephone system to forward the call to a default telephone number.

38. A server for managing a telecommunications system that includes a computer system having a plurality of network logon devices associated with a called party and that includes a telephone system, the server operable to:

a) receive information from the telephone system regarding an incoming telephone call directed to the called party;

b) identify the called party from the information;

c) determine whether one of the network logon devices is logged onto the computer network;

d) if one of the network logon devices is logged onto the computer network, then identify a telephone number associated with the logged-on network logon device and control the telephone system to forward the call to the telephone number; and

e) if none of the network logon devices is logged onto the computer network and if the most recently logged-on network logon device has been logged off the computer network for no longer than a predetermined time, then direct the call to a telephone number associated with the most recently logged-on network logon device.

39. A server for managing a telecommunications system that includes a telephone system, the server operable to:

a) receive information from the telephone system regarding an incoming telephone call from a calling party, the call directed to a called party;

b) present the calling party with a menu listing a plurality of locations to which the call can be forwarded;

c) receive from the calling party a selection signal identifying the location to which the call is to be forwarded; and

d) control the telephone system to forward the call to a telephone number associated with the identified location.

40. A server for managing a telecommunications system that includes a telephone system, the server operable to:

a) receive an indication that the telephone system is to forward calls directed to a first communications device to a second communications device;

b) receive information from the telephone system regarding an incoming telephone call from a calling party, the call directed to the first communications device;

c) record a calling-party message from the calling party;

d) control the telephone system to forward the calling-party message to the second communications device;

e) play the calling-party message at the second communications device;

f) receive a selection signal from the second communications device indicating that the call is to be forwarded to a third communications device; and

g) control the telephone system to forward the call to the third communications device.

41. A server for managing a telecommunications system that includes a computer system having a plurality of network logon devices associated with a called party, a telephone system, and a voice messaging system coupled to the telephone system, the server operable to:

a) receive information from the telephone system regarding an incoming telephone call directed to the called party;

b) determine the called party's telephone number to which the call is to be directed;

c) simulate a "ring no answer" of the telephone number by sending a "ring no answer" signal to the telephone system;

d) intercept control signals provided by the telephone system to the voice messaging system in response to the "ring no answer" signal;

e) record the intercepted control signals;

f) determine whether one of the network logon devices is logged onto the computer network;

g) if no network logon device is logged-on, then transfer the call to the voice messaging system by playing the recorded control signals to the voice messaging system; and

h) if a network logon device is logged-on, then identify a forwarding telephone number associated with the network logon device and control the telephone system to forward the call to the forwarding telephone number.

* * * * *

EXHIBIT 16

REDACTED IN ITS ENTIRETY